



Avaya Solution & Interoperability Test Lab

Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Trunk Emulation – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Empirix Hammer IP test system with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunk emulation. Hammer IP is a test solution for understanding how IP-based systems will behave in the real world. 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, Hammer IP emulates SIP trunks that interface to Session Manager and originates and terminates calls through Avaya SIP telephony network. 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. The 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 the Empirix Hammer IP test system with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunk emulation. Hammer IP is a test solution for understanding how IP-based systems will behave in the real world. 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, Hammer IP emulates SIP trunks that interface to Session Manager and originates and terminates calls through Avaya SIP telephony network. 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. The 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 SIP 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.

- *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 using H.323 Endpoint Emulation [4]*
- *Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager using H.323 Trunk Emulation [5]*

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 SIP trunk channels on Hammer IP and establishing the calls through the Avaya SIP telephony network. The compliance test also covered monitoring various reports on the Hammer IP during and after the test runs, and checking the status of various SIP resources on Communication Manager. The serviceability testing focused on verifying the ability of the Hammer IP to recover from adverse conditions, such as disconnecting the Ethernet cable and rebooting the server.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1 Interoperability Compliance Testing

The interoperability compliance testing focused on verifying that the Hammer IP can establish SIP trunks with Avaya Aura® Session Manager, establish calls, send voice media, and provide voice quality metrics. The following features and functionality were covered:

- Establishing SIP trunks to Avaya Aura® Session Manager and verifying the exchange of SIP Options messages.
- Originating and terminating calls through Avaya SIP telephony network.
- Support of G.711 mu-law and G.729 codecs.
- Support of direct IP-to-IP media (also known as “Shuffling” which allows IP endpoints to send audio RTP packets directly to each other without using media resources on the Avaya Media Gateway). Calls with Shuffling and IP Audio Hairpinning disabled were also verified.
- Generating voice quality metrics with Shuffling disabled.
- DTMF support.
- Exchange of SIP Options messages.
- Originating calls from SIP trunks and terminating calls on SIP trunks, SIP endpoints, H.323 endpoints, and H.323 trunks.

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

2.2 Test Results

All test cases passed. Empirix Hammer IP was successful in originating calls using SIP trunk emulation and terminating calls on channels emulating SIP trunks, SIP endpoints, H.323 trunks, and H.323 endpoints.

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

- The **Guard Time** and **Stagger** parameters may be set too aggressively (e.g., Hammer IP may be initiating too many calls too quickly) and the configuration under test may not be able to handle the load generated by Hammer IP. These parameters should be considered carefully for each test. It may be necessary to slow down the test to a rate that can be reasonably handled by the test configuration.
- Resources may be getting exhausted in the Avaya media gateway. These resources may include media processing resources, touch-tone receivers (TTRs), network trunks, and TDM bus resources.

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

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

2.3 Support

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

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

3 Reference Configuration

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

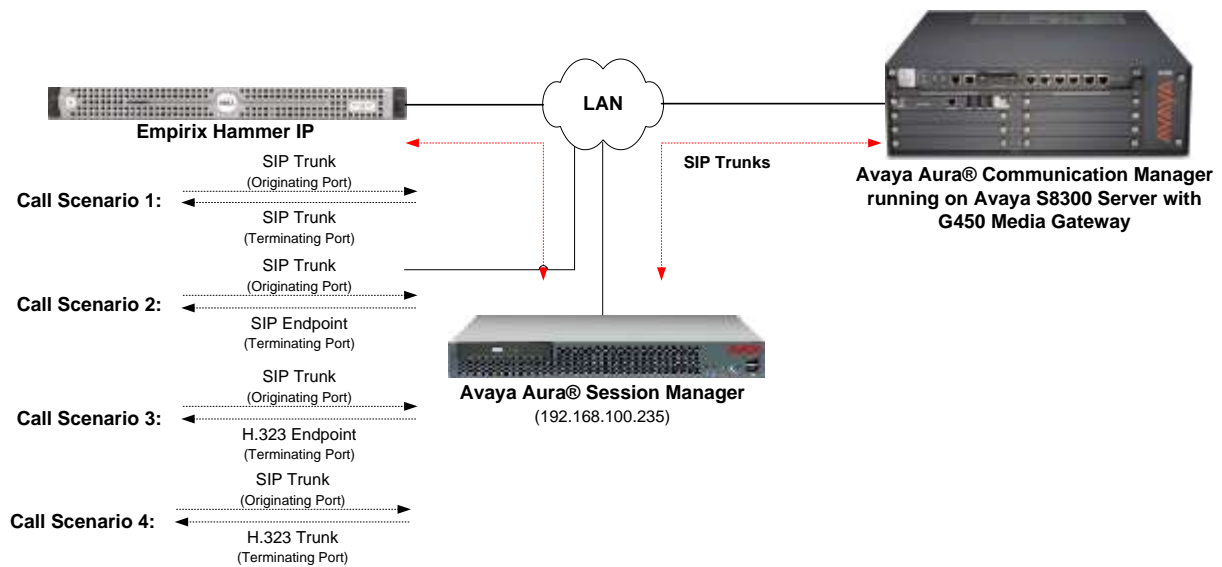


Figure 1: Empirix Hammer IP with Avaya SIP Telephony Network

4 Equipment and Software Validated

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

Equipment	Software
Avaya Aura® Communication Manager running on S8300 Server	6.3.9 SP 9.1 (R016x.03.0.124.0 with Patch 22098)
Avaya G450 Media Gateway	Gateway FW 36.12.0
Avaya Aura® Session Manager running on an S8800 Server	6.3.11.0.631103
Avaya Aura® System Manager	6.3.11 Build No. 6.3.0.8.5682-6.3.84751 Software Update Revision No: 6.3.11.8.2933
Empirix Hammer IP running on Microsoft Windows Server 2008 R2 with Dual 2.40 GHz Intel Xeon CPU and 12.0 GB of RAM	6.0.0.85

5 Configure Avaya Aura® Communication Manager

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

- Administer IP Node Names
- Administer IP Codec Set
- Administer IP Network Region
- Administer SIP Trunk Group to Session Manager
- Administer AAR Call Routing

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

5.1 Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for the S8300 Server in the G450 Media Gateway (*procr*) and Session Manager (*lz-asm*). 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
devcon13                 10.32.24.20
lz-asm                  192.168.100.235
procr                   192.168.100.10
procr6                   ::
( 5 of 5 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 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.711 mu-law.

```
change ip-codec-set 1                                     Page 1 of 2
                                                    IP Codec Set
      Codec Set: 1
      Audio      Silence      Frames      Packet
      Codec      Suppression  Per Pkt    Size (ms)
1:  G.711MU      n            2        20
2:
3:
```

5.3 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. The **Authoritative Domain** for this configuration is *devcon.com*.

change ip-network-region 1		Page 1 of 20
IP NETWORK REGION		
Region: 1		
Location: 1		Authoritative Domain: devcon.com
Name:		
MEDIA PARAMETERS		Intra-region IP-IP Direct Audio: no
Codec Set: 1		Inter-region IP-IP Direct Audio: no
UDP Port Min: 2048		IP Audio Hairpinning? n
UDP Port Max: 65535		
DIFFSERV/TOS PARAMETERS		
Call Control PHB Value: 46		
Audio PHB Value: 46		
Video PHB Value: 26		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 6		
Audio 802.1p Priority: 6		
Video 802.1p Priority: 5		
AUDIO RESOURCE RESERVATION PARAMETERS		
H.323 IP ENDPOINTS		RSVP Enabled? n
H.323 Link Bounce Recovery? y		
Idle Traffic Interval (sec): 20		
Keep-Alive Interval (sec): 5		
Keep-Alive Count: 5		

5.4 Administer SIP Trunk to Session Manager

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tcp*.
- Specify the S8300 and the Session Manager 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 *5060* 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.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *devcon.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.
- Disable **Initial IP-IP Direct Media**.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

add signaling-group 60		Page 1 of 3
SIGNALING GROUP		
Group Number: 60	Group Type: sip	
IMS Enabled? n	Transport Method: tcp	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: SM	
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y		
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n		
Alert Incoming SIP Crisis Calls? n		
Near-end Node Name: procr	Far-end Node Name: lz-asm	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 1	
Far-end Domain: devcon.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
	RFC 3389 Comfort Noise? n	
DTMF over IP: rtp-payload	Direct IP-IP Audio Connections? y	
Session Establishment Timer(min): 3	IP Audio Hairpinning? n	
Enable Layer 3 Test? y	Initial IP-IP Direct Media? n	
H.323 Station Outgoing Direct Media? n	Alternate Route Timer(sec): 6	

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to Hammer IP. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Accept the default values for the remaining fields.

add trunk-group 60		Page 1 of 21	
TRUNK GROUP			
Group Number: 60	Group Type: sip	CDR Reports: y	
Group Name: To lz-asm	COR: 1	TN: 1	TAC: 1060
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Member Assignment Method: auto	
		Signaling Group: 60	
		Number of Members: 40	

5.5 AAR Call Routing

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and enter add an entry that routes digits beginning with “46” to route pattern 60 as shown below. Note that the digits arriving from Hammer IP for incoming call requests will start with ‘8’, the AAR feature access code. The AAR feature access code will steer the call to AAR routing.

change aar analysis 4						Page 1 of 2	
AAR DIGIT ANALYSIS TABLE							
Location: all						Percent Full: 2	
	Dialed String	Total		Route	Call	Node	ANI
		Min	Max	Pattern	Type	Num	Reqd
46		5	5	60	aar		n
5		7	7	254	aar		n
6		5	5	2	aar		n
7		5	5	3	aar		n
8		7	7	254	aar		n

Configure a preference in **Route Pattern** 60 to route calls over SIP trunk group 60 as shown below.

change route-pattern 60		Page 1 of 3	
Pattern Number: 60		Pattern Name: To lz-asm	
SCCAN? n		Secure SIP? n	
Grp No	FRL NPA Pfx Hop Toll No.	Inserted Dgts	DCS/ IXC QSIG Intw
1: 60	0		n user
2:			n user
3:			n user
4:			n user
5:			n user
6:			n user

6 Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities corresponding to Session Manager, Communication Manager, and Hammer IP
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies and Dial Patterns
- Session Manager, corresponding to the Avaya Aura® Session Manager Server to be managed by Avaya Aura® System Manager

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager using the URL “https://<ip-address>/SMGR”, where <ip-address> is the IP address of Avaya Aura® System Manager. Log in with the appropriate credentials.

6.1 Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button on the right (not shown). The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., *devcon.com*).
- **Type:** Set to *SIP*.
- **Notes:** Descriptive text (optional).

Click **Commit**.

Since the sample configuration does not deal with any other domains, no additional domains need to be added.

The screenshot shows the Avaya Aura System Manager 6.3 GUI. The left sidebar contains a navigation menu with options: Home, Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Domain Management' and displays a table with one item. The table has columns for Name, Type, and Notes. The item listed is 'devcon.com' with Type 'SIP'. There are 'Commit' and 'Cancel' buttons at the top right and bottom right of the table area. The top of the GUI shows the Avaya logo and 'Aura System Manager 6.3'. The top right corner indicates 'Last Logged on: At March 23, 2015 4:30 PM' and a 'Log off admin' link.

Name	Type	Notes
devcon.com	SIP	

6.2 Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

The screen below shows addition of the *Lincroft* location, which includes the Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

Under *Location Pattern*:

- **IP Address Pattern:** A pattern used to logically identify the location.
- **Notes:** Descriptive text (optional).

Click **Commit** to save the **Location** definition.

6.3 Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager, the S8300 Server in the G450 Media Gateway, and Hammer IP.

6.3.1 Avaya Aura® Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

The screenshot displays the Avaya Aura System Manager 6.3 web interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 6.3', and a user session summary: 'Last Logged on at March 23, 2015 6:20 PM' with a 'Log off admin' link. The left sidebar contains a menu with 'Routing' selected, and sub-items: Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area shows the breadcrumb 'Home / Elements / Routing / SIP Entities' and a 'Help ?' link. The 'SIP Entity Details' form is titled 'General' and includes fields for: Name (labeled with a red asterisk, value: 'lz-asm'), FQDN or IP Address (labeled with a red asterisk, value: '192.168.100.235'), Type (dropdown menu showing 'Session Manager'), Notes (text area), Location (dropdown menu showing 'Lincroft'), Outbound Proxy (dropdown menu), Time Zone (dropdown menu showing 'America/New_York'), and Credential name (text area). At the bottom, the 'SIP Link Monitoring' section has a dropdown menu set to 'Use Session Manager Configuration'. 'Commit' and 'Cancel' buttons are located at the top right of the form area.

Under *Port*, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain** The domain used for the enterprise (e.g., *devcon.com*).

Defaults can be used for the remaining fields. Click **Commit** (not shown) to save the SIP Entity definition.

The screenshot shows a web interface for configuring SIP entities. At the top, there are fields for 'TCP Failover port' and 'TLS Failover port', each with an 'Add' and 'Remove' button. Below these is a table titled '3 Items' with a 'Filter: Enable' link. The table has five columns: a checkbox, 'Port', 'Protocol', 'Default Domain', and 'Notes'. There are three rows in the table. The second row, which has '5060' in the 'Port' column, 'UDP' in the 'Protocol' column, and 'devcon.com' in the 'Default Domain' column, is highlighted with a red rectangle. The 'Notes' column for this row is empty. At the bottom of the table, there is a 'Select' dropdown menu with 'All' and 'None' options.

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	devcon.com	
<input type="checkbox"/>	5060	UDP	devcon.com	
<input type="checkbox"/>	5061	TLS	devcon.com	

6.3.2 Avaya Aura® Communication Manager

A SIP Entity must be added for the Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., S8300 Server) on the telephony system.
- **Type:** Select *CM*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

The screenshot displays the Avaya Aura System Manager 6.3 web interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 6.3', and a user session summary showing 'Last logged on at March 23, 2015 4:10 PM' and a 'Log off admin' link. The left sidebar contains a tree view with the following items: Home, Routing, Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area shows the breadcrumb 'Home / Elements / Routing / SIP Entities' and the title 'SIP Entity Details'. Below the title is the 'General' tab. The form contains the following fields: 'Name' (text box with 'devcon14'), 'FQDN or IP Address' (text box with '192.168.100.10'), 'Type' (dropdown menu with 'CM' selected), 'Notes' (text box), 'Adaptation' (dropdown menu), 'Location' (dropdown menu with 'Lincoln' selected), 'Time Zone' (dropdown menu with 'America/New_York' selected), 'SIP Timer B/F (in seconds)' (text box with '4'), 'Credential name' (text box), and 'Call Detail Recording' (dropdown menu with 'none' selected). At the top right of the form area are 'Commit' and 'Cancel' buttons, and a 'Help ?' link.

6.3.3 Empirix Hammer IP

Two SIP Entities must be added for Empirix Hammer IP, one for incoming calls and another one for outgoing calls. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., *Hammer-Inc*) on the telephony system.
- **Type:** Select *SIP Trunk*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

The following SIP Entity is for incoming call requests from Hammer IP.

The screenshot displays the Avaya Aura System Manager 6.3 web interface. The left-hand navigation pane shows a tree structure with 'Routing' selected, and 'SIP Entities' highlighted under the 'Routing' category. The main content area is titled 'SIP Entity Details' and shows the 'General' tab. The form contains the following fields and values:

- Name:** Hammer-Inc
- FQDN or IP Address:** 192.168.100.170
- Type:** SIP Trunk
- Notes:** (empty)
- Adaptation:** (empty)
- Location:** Lincroft
- Time Zone:** America/New_York
- SIP Timer B/F (in seconds):** 4
- Credential name:** (empty)
- Call Detail Recording:** egress
- Loop Detection Mode:** Off
- SIP Link Monitoring:** Use Session Manager Configuration

Buttons for 'Commit' and 'Cancel' are visible at the top right of the form area. The top of the page shows the user is logged in as 'admin' on April 17, 2015.

The following SIP Entity is for calls destined to Hammer IP.

The screenshot displays the Avaya Aura System Manager 6.3 web interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 6.3', and a user status area indicating 'Last logged on at April 17, 2015 4:14 PM' with 'Log off admin' and 'On Duty' buttons. A left-hand menu lists various configuration areas: Routing, Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Home / Elements / Routing / SIP Entities' and contains the 'SIP Entity Details' form for the entity named 'Hammer-Out'. The form includes fields for Name, FQDN or IP Address (192.168.100.171), Type (SIP Trunk), Notes, Adaptation, Location (Lincoln), Time Zone (America/New_York), SIP Timer B/F (4), Credential name, Call Detail Recording (egress), Loop Detection Mode (Off), and SIP Link Monitoring (Use Session Manager Configuration). 'Commit' and 'Cancel' buttons are located at the top right of the form area.

AVAYA
Aura System Manager 6.3

Last logged on at April 17, 2015 4:14 PM
On Duty Log off admin

Home Routing

Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

Home / Elements / Routing / SIP Entities

SIP Entity Details

General

Name: Hammer-Out

FQDN or IP Address: 192.168.100.171

Type: SIP Trunk

Notes:

Adaptation:

Location: Lincoln

Time Zone: America/New_York

SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: egress

Loop Detection

Loop Detection Mode: Off

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

Commit Cancel

6.4 Add Entity Links

This section covers the configuration of Entity Links for Communication Manager and Hammer IP.

6.4.1 Communication Manager Entity Link

The SIP trunk from Session Manager to Communication Manager is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *lz-asm to devcon14*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Select *Trusted*. *Note: If Trusted is not selected, calls from the associated SIP Entity specified in Section 6.3.2 will be denied.*

Click **Commit** to save the Entity Link definition.

The following Entity Link is between Session Manager and Communication Manager.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy	Deny New Service	Notes
lz-asm to devcon14	lz-asm	TCP	5060	devcon14	No	5060	Trusted		

6.4.2 Hammer IP Entity Links

The SIP trunk from Session Manager to Hammer IP is described by an Entity link. Two entity links are required for Hammer IP, one for incoming calls from Hammer IP and another one for outgoing calls to Hammer IP. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *Hammer-Inc Link* or *Hammer-Out Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the *Hammer-Inc* or *Hammer-Out* SIP entity.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Selected *Trusted*. *Note: If the link is not trusted, calls from the associated SIP Entity specified in Section 6.3.3 will be denied.*

Click **Commit** to save the Entity Link definition.

The following Entity Link is between Session Manager and the SIP Entity that handles incoming calls from Hammer IP (i.e., *Hammer-Inc*).



The following Entity Link is between Session Manager and the SIP Entity that handles outgoing calls to Hammer IP (i.e., *Hammer-Out*).

The screenshot shows the Avaya Aura System Manager 6.3 interface. The left sidebar contains a navigation menu with the following items: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Entity Links' and shows a breadcrumb path: Home / Elements / Routing / Entity Links. Below the title is a table with the following columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, DRS Override, Port, Connection Policy, Degr. Res. Service, and Notes. A single entry is listed in the table:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DRS Override	Port	Connection Policy	Degr. Res. Service	Notes
* Hammer-Out	* 1p-asm	UDP	* 5060	* Hammer-Out	<input type="checkbox"/>	* 5060	trusted		

At the bottom of the table, it says 'Selected: All items'. There are 'Commit' and 'Cancel' buttons at the top right and bottom right of the configuration area.

6.5 Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.3**. Two routing policies were added – one for Communication Manager and one for Hammer IP. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:

Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screen shows the Routing Policy for Communication Manager.

The screenshot displays the Avaya Aura System Manager 8.3 interface. The left sidebar shows a navigation menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and contains the following sections:

- General:** Includes fields for 'Name' (devcon14 Policy), 'Disabled' (checkbox), 'Retries' (0), and 'Notes'.
- SIP Entity as Destination:** Includes a 'Select' button and a table listing available SIP entities.
- Time of Day:** Includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons, and a table for defining time ranges.

Name	FQDN or IP Address	Type	Notes
devcon14	192.168.100.10	CM	

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	12	12	12	12	12	12	12	00:00	23:59	Time Range 24/7

The following screen shows the Routing Policy for Hammer IP.

The screenshot displays the Avaya Aura System Manager 6.3 web interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies (selected), Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Home / Elements / Routing / Routing Policies'. It features a 'Routing Policy Details' section with 'Commit' and 'Cancel' buttons. Below this is the 'General' tab, which includes fields for 'Name' (set to 'Hammer Policy'), 'Disabled' (checkbox), 'Retries' (set to 0), and 'Notes'. The 'SIP Entry as Destination' section has a 'Select' button and a table with columns 'Name', 'FQDN or IP Address', 'Type', and 'Notes'. The table contains one entry: 'Hammer-Out' with IP '192.168.100.171' and Type 'SIP Trunk'. The 'Time of Day' section includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. It shows a table with columns for 'Ranking', 'Name', and days of the week (Mon-Sun), along with 'Start Time', 'End Time', and 'Notes'. One entry is visible: '24/7' with a start time of '00:00' and end time of '23:59'. The bottom of the interface shows a 'Select: All, None' option.

AVAYA
Aura System Manager 6.3

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit Cancel

General

* Name: Hammer Policy

Disabled: ☐

* Retries: 0

Notes:

SIP Entry as Destination

Select

Name	FQDN or IP Address	Type	Notes
Hammer-Out	192.168.100.171	SIP Trunk	

Time of Day

Add Remove View Gaps/Overlaps

1 Item

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select: All, None

6.6 Add Dial Patterns

Dial patterns must be defined to direct calls to the appropriate SIP Entity. In the sample configuration, a 6-digit number beginning with '8' followed by "46202" will be routed to Communication Manager. The '8' is the AAR access code and "46202" are the digits routed to the Hammer IP, which will terminate on a SIP trunk. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under *General*:

- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern (optional).

Under *Originating Locations and Routing Policies*:

Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definition for Communication Manager. Based on these digits, Communication Manager will route the call to Hammer IP via a SIP trunk.

The screenshot shows the Avaya Aura System Manager 11.2 interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns (selected), Regular Expressions, and Defaults. The main content area is titled 'Dial Pattern Details' and includes a 'General' tab. The form fields are as follows:

- Pattern: 846
- Min: 6
- Max: 6
- Emergency Call: ☐
- Emergency Priority: 1
- Emergency Type:
- SIP Domain: devcon.com
- Notes:

Below the form is a section titled 'Originating Locations and Routing Policies' with 'Add' and 'Remove' buttons. It contains a table with one entry:

Originating Location Name	Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
Unicraft	DevCon2 network	Devcon14 Policy	1	<input type="checkbox"/>	Devcon14	

At the bottom of the table, there is a 'Select: All, None' link.

The following screen shows the dial pattern definition for Hammer IP. The extension, 46202, will be routed to Hammer IP. Note that “46202” does not have to match any configuration on Hammer IP. Hammer IP will answer any calls routed to it regardless of the digits.

Avaya
Aura® System Manager 6.3

Home / Elements / Routing / Dial Patterns

Dial Pattern Details [Cancel] [Cancel]

General

* Pattern: 462

* Min: 5

* Max: 5

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: devcon.com

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
Lincroft	DevConnet network	Hammer Policy	5	<input type="checkbox"/>	Hammer-Out	

Select: All, None

6.7 Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add** (not shown), and fill in the fields as described below and shown in the following screen:

Under *General*:

- **SIP Entity Name:** Select the name of the SIP Entity added for Session Manager
- **Description:** Descriptive comment (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

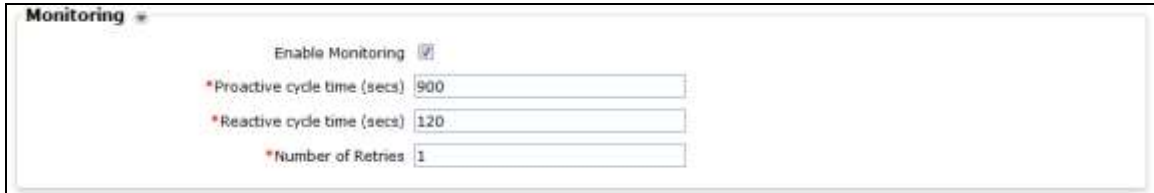
Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager

Use default values for the remaining fields. Click **Commit** to add this Session Manager.

The screenshot shows the Avaya Aura System Manager 6.3 web interface. The left-hand navigation menu is expanded, showing the 'Session Manager' section. The 'Session Manager Administration' option is selected. The main content area displays the 'Edit Session Manager' configuration page. The page has a breadcrumb trail: 'Home / Elements / Session Manager / Session Manager Administration'. There are 'Commit' and 'Cancel' buttons at the top right of the configuration area. The configuration is divided into two sections: 'General' and 'Security Module'. The 'General' section includes fields for 'SIP Entity Name' (l2-sm), 'Description', 'Management Access Point Host Name/IP' (192.168.100.233), 'Direct Routing to Endpoints' (Enable), and 'VMware Virtual Machine'. The 'Security Module' section includes fields for 'SIP Entity IP Address' (192.168.100.233), 'Network Mask' (255.255.255.0), 'Default Gateway' (192.168.100.1), 'Call Control PHB' (46), 'QoS Priority' (6), 'Speed & Duplex' (Auto), 'VLAN ID', and 'SIP Firewall Configuration' (SM 6.3.8.0). The top right corner shows the user's login information: 'Last Logged on at March 23, 2015 4:20 PM' and a 'Log off admin' button.

The following screen shows the **Monitoring** section, which determines how frequently Session Manager sends SIP Options messages to Hammer IP. Use default values for the remaining fields. Click **Commit** to add this Session Manager. In the following configuration, Session Manager sends a SIP Options message every 900 secs. If there is no response, Session Manager will send a SIP Options message every 120 secs.



The image shows a configuration window titled "Monitoring" with a close button (X). Inside the window, there are three settings:

- "Enable Monitoring" with a checked checkbox.
- "* Proactive cycle time (secs)" with a text input field containing the value "900".
- "* Reactive cycle time (secs)" with a text input field containing the value "120".
- "* Number of Retries" with a text input field containing the value "1".

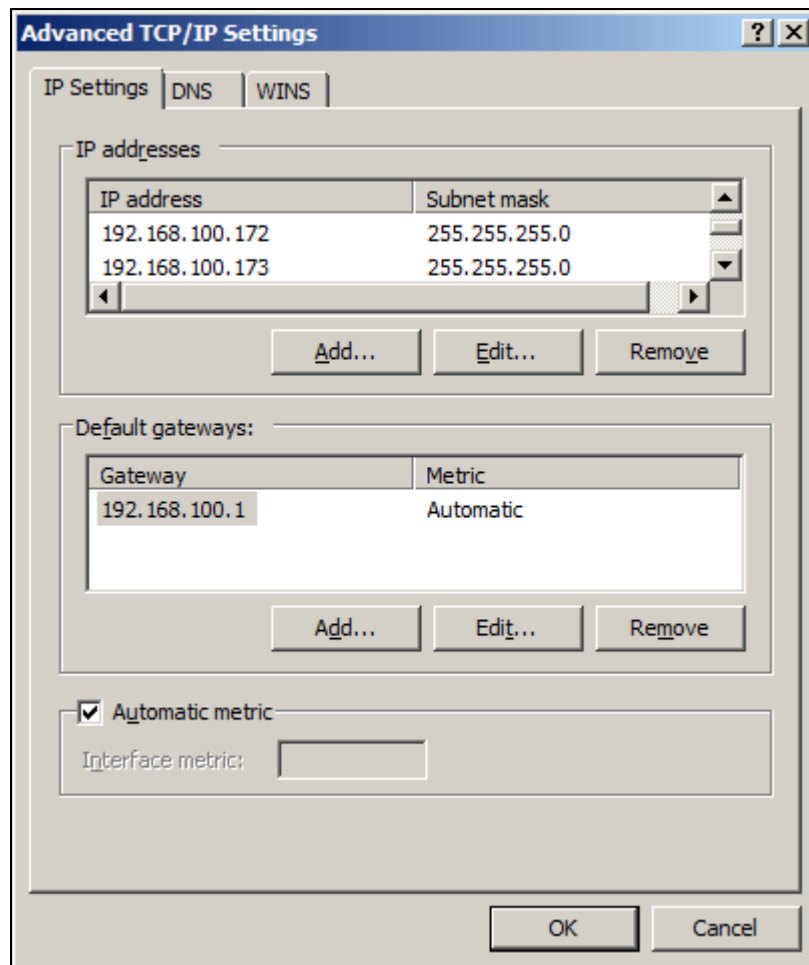
7 Configure Empirix Hammer IP

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 PhoneBook, using the **Hammer Configurator**.
- Save and apply the Hammer configuration and start the Hammer server.
- Create and run the test script using the **Hammer TestBuilder**.

7.1 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 SIP trunk 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 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.



7.2 Configure System

This section covers the configuration of originating and terminating channels. In this configuration, the originating channels emulate SIP trunks (described in **Section 7.2.1**). The terminating channels can emulate SIP trunks, H.323 trunks, SIP endpoints or H.323 endpoints. These Application Notes will explicitly describe the configuration for terminating calls to SIP trunks in **Section 7.2.2.1**. In addition, it will provide references to other Application Notes for configuring terminating channels as SIP endpoints, H.323 endpoints, and H.323 trunks in **Sections 7.2.2.2, 7.2.2.3, and 7.2.2.4**, respectively. Only one of those sections needs to be followed depending on the configuration desired.

7.2.1 Configure Originating Channels – SIP Trunks

The 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.

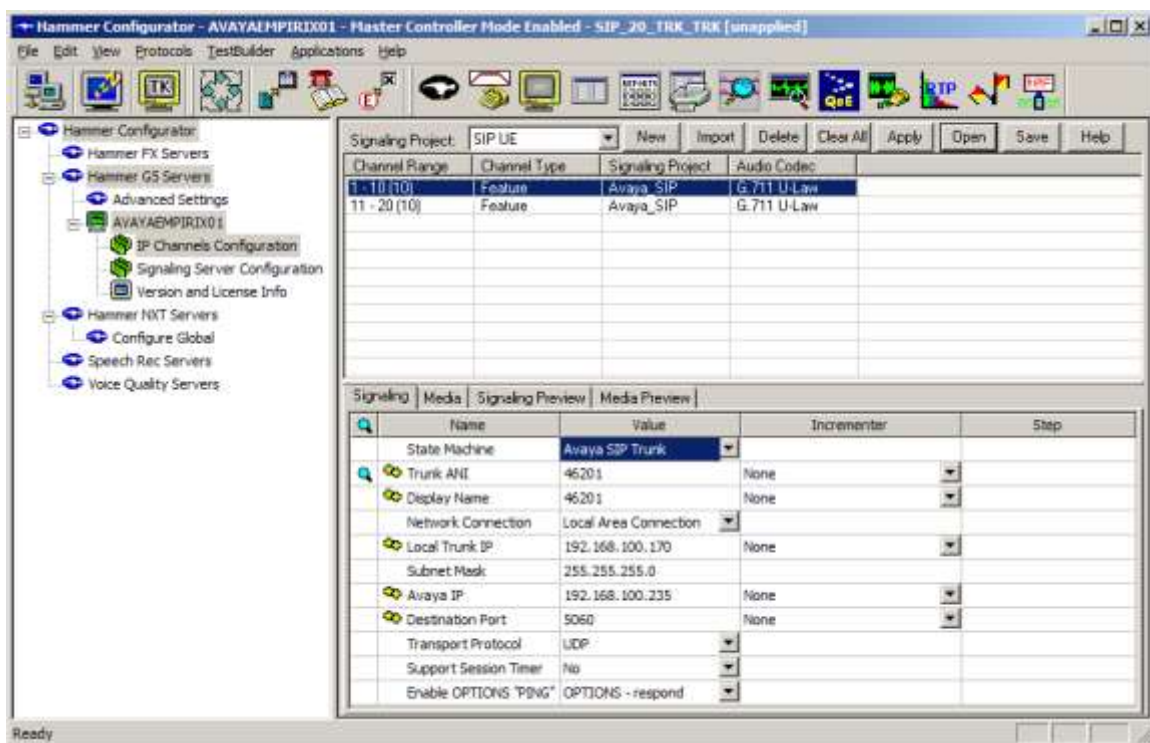
Note: It is assumed that Hammer IP is already in **Master Controller Mode**. To verify, check that the title bar of the **Hammer Configurator** indicates *Master Controller Mode Enabled* as shown below. It is also assumed that a system was already added to the configuration. In this configuration, the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



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

- **State Machine** should be set to *Avaya SIP Trunk*.
- **Trunk ANI** may be any extension.
- **Display Name** may be any extension.
- **Network Connection** should be set to the appropriate network interface.
- **Local Trunk IP** should be set to a unique IP address (e.g., *192.168.100.170*) and should match the IP address configured on Communication Manager in **Sections 5.1** and **5.4**. This IP address will be used for the group of originating channels.
- **Subnet Mask** should be set to the network mask (e.g., *255.255.255.0*).
- **Avaya IP** should be set to the Session Manager SIP interface (e.g., *192.168.100.235*).
- **Destination Port** should be set to the SIP listen port (e.g., *5060*).
- **Transport Protocol** should be set to *UDP*.
- **Enable OPTIONS “PING”** should be set to *OPTIONS - respond*.
- The default values for other fields may be used as shown.



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 channel group (e.g., 192.168.100.170).
- **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.

Hammer Configurator - AVAYAEMPIRX01 - Master Controller Mode Enabled - SIP_20_TRK_TRK [unapplied]

File Edit View Protocols TestBuilder Applications Help

Hammer Configurator

- Hammer FX Servers
- Hammer G5 Servers
 - Advanced Settings
 - AVAYAEMPIRX01
 - IP Channels Configuration
 - Signaling Server Configuration
 - Version and License Info
- Hammer NXT Servers
 - Configure Global
 - Speech Rec Servers
 - Voice Quality Servers

Signaling Project: SIP UE New Import Delete Clear All Apply Open Save Help

Channel Range	Channel Type	Signaling Project	Audio Codec
1 - 10 (10)	Feature	Avaya SIP	G.711 U-Law
11 - 20 (10)	Feature	Avaya SIP	G.711 U-Law

Signaling Media Signaling Preview Media Preview

Name	Value	Incrementer	Step
Audio Codec	G.711 U-Law		
Frequency [ms]	20 [ms]		
Network Connection	Local Area Connection		
Source IP Address	192.168.100.170	None	
Audio Port	10000	++++++	2
DTMF Type	In Band		
Silence Type	Audio		
Jitter Buffer	8 x Frequency [ms]		
Subnet Mask	255.255.255.0		
Media Profile	G711U.sdp		
Video Xmit Codec	Automatic		
T38 Port	20000	++++++	2
Video Port	30000	++++++	2
Video Silence	Gap		
RTCP	Enabled		
TestBuilder Configuration	None		
SRTP Encryption	Disabled		
SRTP Authentication	Disabled		
SDES	Disabled		
Fax Pass-Through	Disabled		

Ready

7.2.2 Configure Terminating Channels

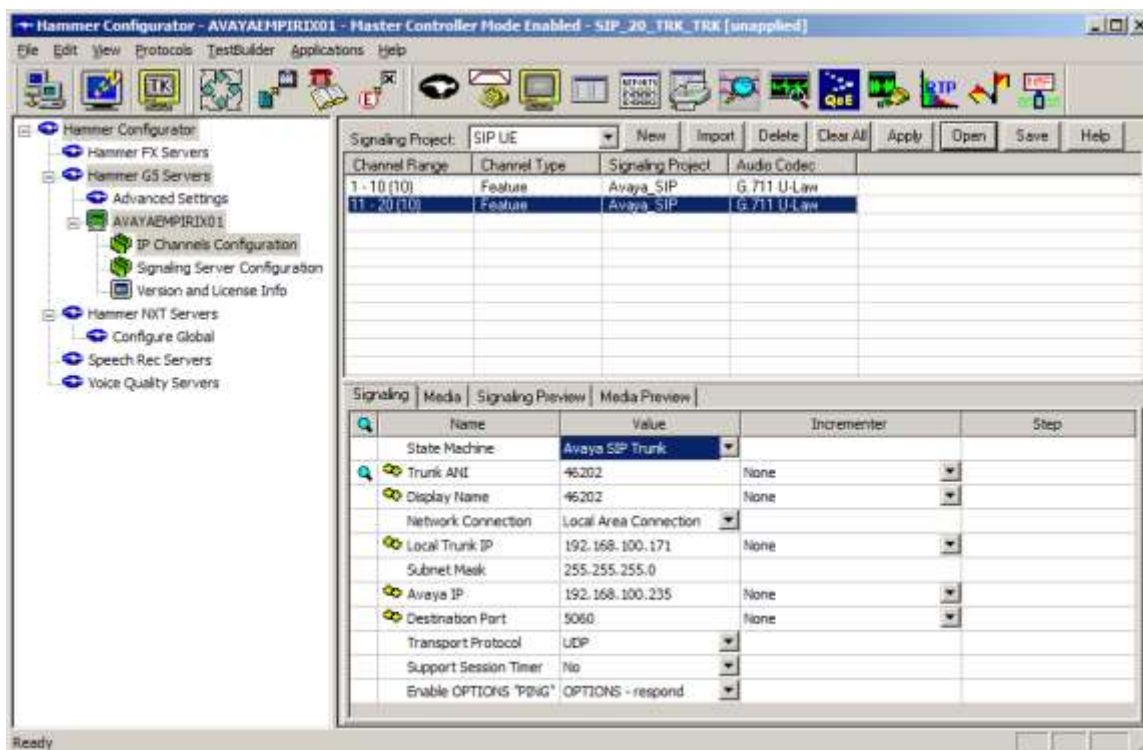
During the compliance test, the originating channels emulated SIP endpoints with the calls terminating on SIP endpoints, SIP trunks, or H.323 endpoints. Select one of the following subsections depending on the configuration desired.

- **Section 7.2.2.1** for terminating calls on SIP trunks.
- **Section 7.2.2.2** for terminating calls on SIP endpoints.
- **Section 7.2.2.3** for terminating calls on H.323 endpoints.
- **Section 7.2.2.4** for terminating calls on H.323 trunks.

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

7.2.2.1 Configure Terminating Channels – SIP Trunks

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 7.2.1** with the exception that the **Trunk ANI**, **Display Name**, and **Local Trunk IP** fields will be different. This group of channels will be assigned IP address *192.168.100.171*.



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

Hammer Configurator - AVAYAEMPIRX01 - Master Controller Mode Enabled - SIP_TRK_TRK [unapplied]

File Edit View Protocols TestBuilder Applications Help

Hammer Configurator

- Hammer FX Servers
- Hammer G5 Servers
 - Advanced Settings
 - AVAYAEMPIRX01
 - IP Channels Configuration
 - Signaling Server Configuration
 - Version and License Info
- Hammer NXT Servers
 - Configure Global
- Speech Rec Servers
- Voice Quality Servers

Signaling Project: SIP_TRK_TRK [unapplied] New Import Delete Clear All Apply Open Save Help

Channel Range	Channel Type	Signaling Project	Audio Codec
1 - 10 (10)	Feature	Avaya SIP	G.711 U-Law
11 - 20 (10)	Feature	Avaya SIP	G.711 U-Law

Signaling Media Signaling Preview Media Preview

Name	Value	Incrementer	Step
Audio Codec	G.711 U-Law		
Frequency [ms]	20 [ms]		
Network Connection	Local Area Connection		
Source IP Address	192.168.100.171	None	
Audio Port	10000	++++++	2
DTMF Type	In Band		
Silence Type	Audio		
Jitter Buffer	8 x Frequency [ms]		
Subnet Mask	255.255.255.0		
Media Profile	G711U.sdp		
Video Xmit Codec	Automatic		
T38 Port	20000	++++++	2
Video Port	30000	++++++	2
Video Silence	Gap		
RTCP	Enabled		
TestBuilder Configura...	None		
SRTP Encryption	Disabled		
SRTP Authentication	Disabled		
SDES	Disabled		
Fax Pass-Through	Disabled		

Ready

7.2.2.2 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 stations and call routing on Communication Manager.
- **Section 6** describes how to configure SIP endpoints on Session Manager.
- **Section 7.2.2.1** describes how to configure terminating SIP endpoints on Hammer IP.
- **Section 7.2.3** describes how to configure the PhoneBook.
- **Section 7.4** describes how to disable the **Do Connect Latency** option (required) and how to specify the dialed digits when running a test script.

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

7.2.2.3 Configure Terminating Channels – H.323 Endpoints

To terminate the calls to H.323 endpoints follow the instructions described in [4], specifically:

- **Section 5** describes how to configure H.323 endpoints for the originating channels on Communication Manager.
- **Section 6.2.2.1** describes how to configure terminating H.323 endpoints on Hammer IP.
- **Section 6.2.3** describes how to configure the PhoneBook.
- **Section 6.4** 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 H.323 endpoints.

7.2.2.4 Configure Terminating Channels – H.323 Trunks

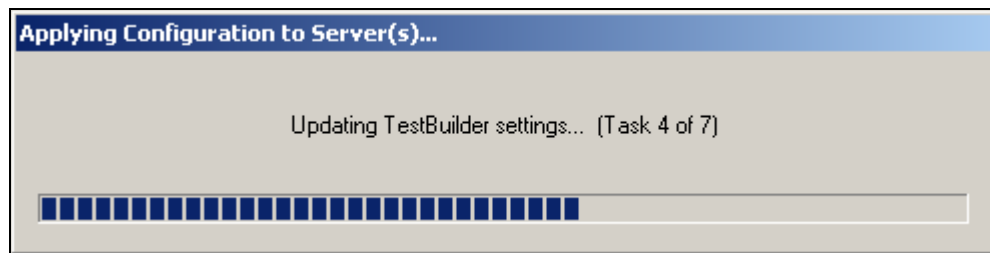
To terminate the calls to H.323 trunks follow the instructions described in [5], specifically:

- **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.

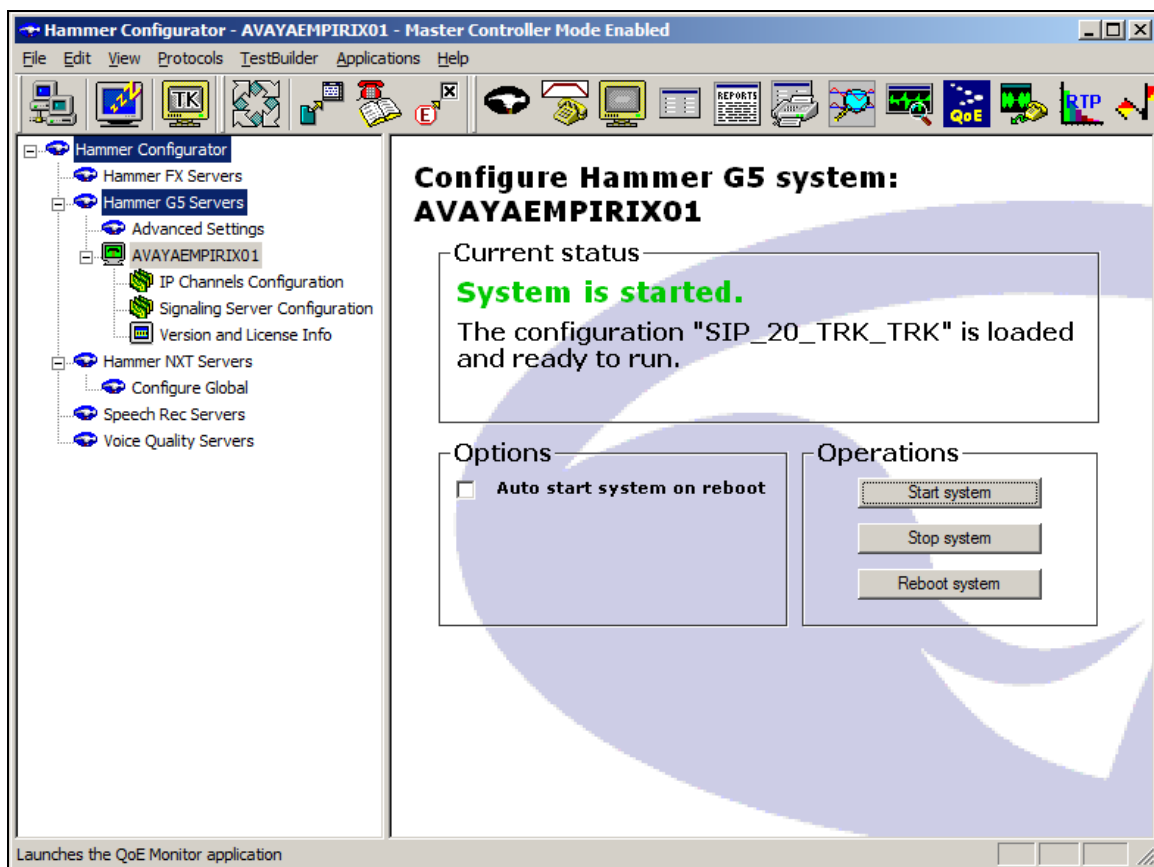
The configuration described in all the aforementioned sections of [5] must be completed for terminating calls to H.323 trunks.

7.3 Applying the Hammer IP Configuration

This completes the configuration of Hammer IP. This configuration should be saved by clicking the **Save** button on the **Hammer Configurator** window. The configuration needs to be applied to the server for the changes to take effect. Click on the **Apply** button in the **Hammer Configurator** window. The following window is displayed as the configuration is being applied to the server.



Check that the system has been started by clicking on the server name (e.g., *AVAYAEMPIRIX01*) in the left pane of the **Hammer Configurator**. If the current status is *System Is Stopped*, click the **Start system** button to start the system. When the system is started, it should appear as shown below and should also specify which configuration has been applied. The configuration performed above was saved as *SIP_20_TRK_TRK*.

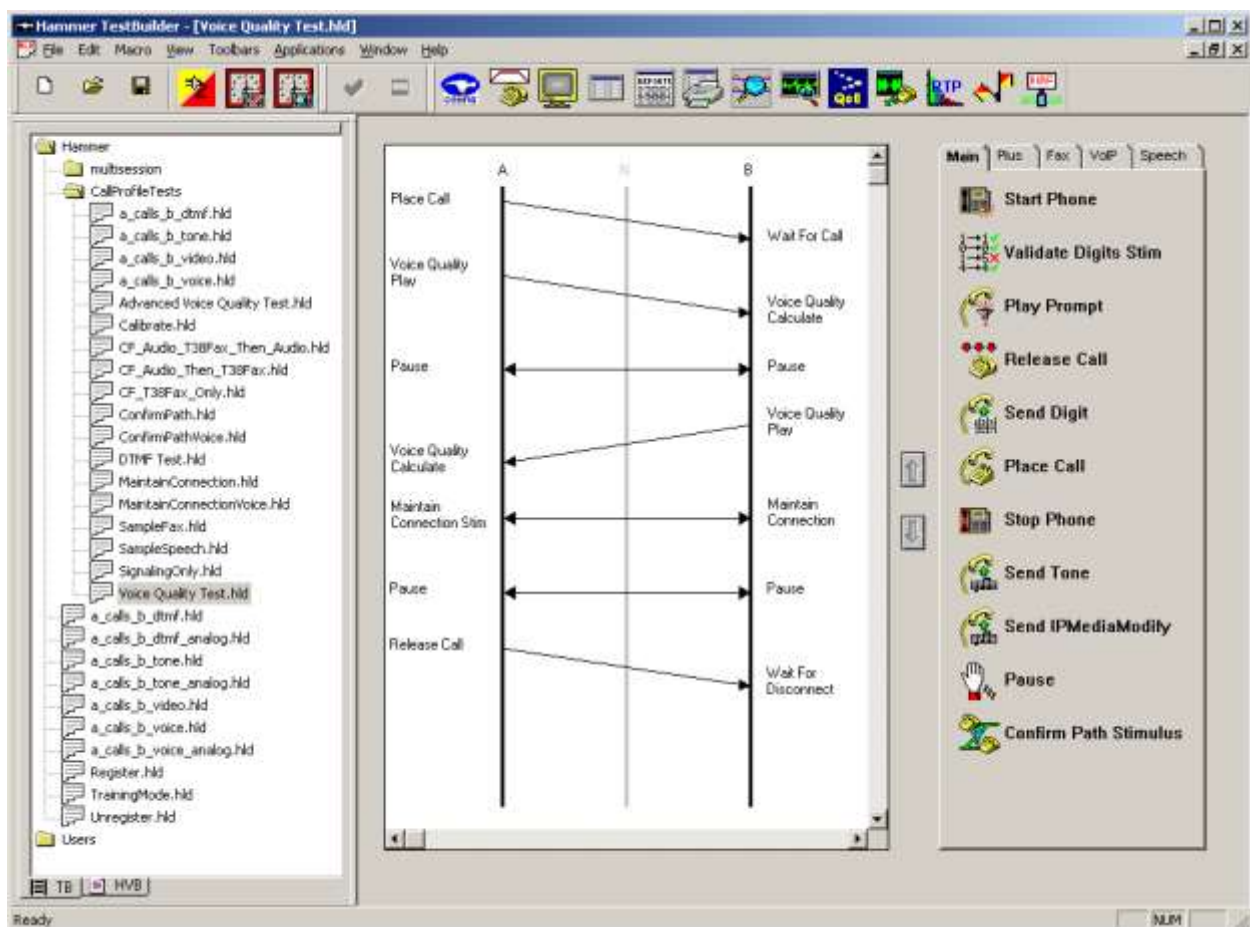


7.4 Configure and Run the Test Script

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

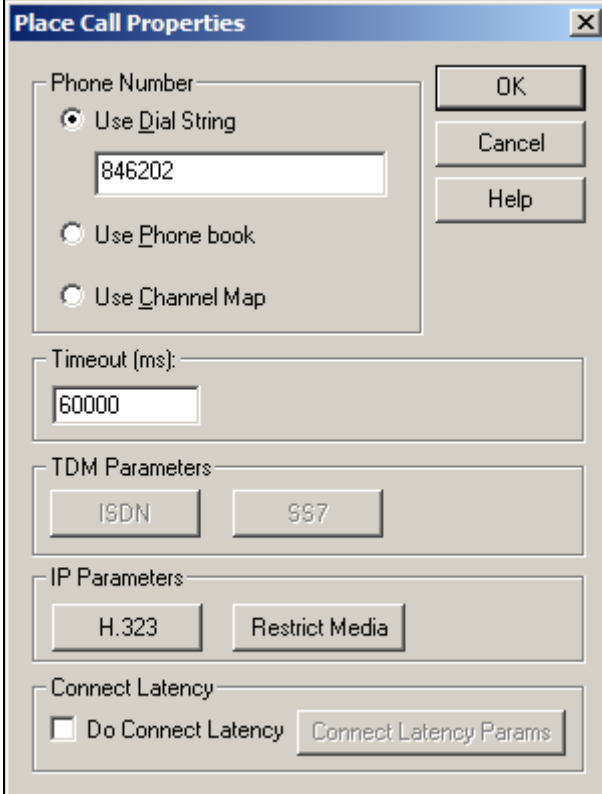
- a_calls_b_dtmf.hld to verify DTMF
- Voice Quality Test.hld to verify voice quality

The sample test script, Voice Quality Test.hld, establishes a VoIP call between two H.323 endpoints on 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 SIP endpoint) places a call to the B-side (terminating SIP 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 dial the same digits for every call. In this example, the Hammer IP dials the AAR access code '8' followed by "46202".

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

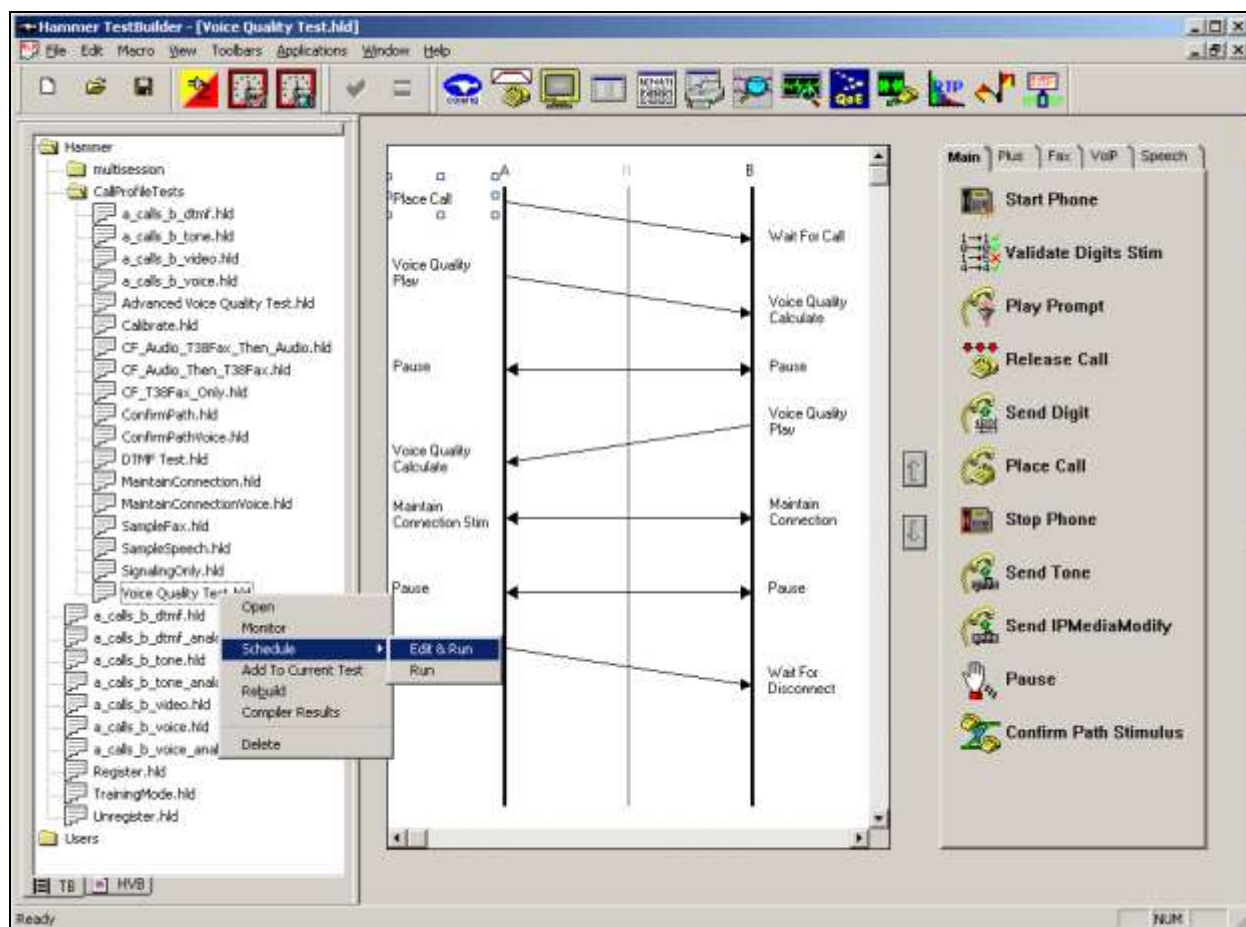


The image shows a "Place Call Properties" dialog box with the following sections and controls:

- Phone Number:** A group box containing three radio buttons: "Use Dial String" (selected), "Use Phone book", and "Use Channel Map". Below the radio buttons is a text field containing "846202".
- Timeout (ms):** A text field containing "60000".
- TDM Parameters:** A group box containing two buttons: "ISDN" and "SS7".
- IP Parameters:** A group box containing two buttons: "H.323" and "Restrict Media".
- Connect Latency:** A group box containing a checkbox labeled "Do Connect Latency" (which is unchecked) and a button labeled "Connect Latency Params".

On the right side of the dialog box, there are three buttons: "OK", "Cancel", and "Help".

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**. 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.

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.

Properties

TB Scheduler | Other

...ary\Hammer\CallProfileTests\Voice Quality Test.hld

Start Time: 4:03:38 PM 4/17/2015

Action if a Channel is busy: Wait

Channels

A-Side: AVAYAEMPIRIX01[1-10] ...

B-Side: AVAYAEMPIRIX01[11-20]

PhoneBook

Select a PhoneBook: Default-phonebook

Stagger

☐ Automatic - Est. CHT (s) 5

☒ User Defined - (ms) 50

☐ Random - Min (s) 1 Max (s) 5

☐ None

Max Active Connections: 0 (0 = Unlimited)

Max Test Time: Hours: 0 Minutes: 0 (0 = Forever)

Loop Count: (-1 = Loop Forever) -1

Guard Time (ms): 3500

OK Cancel Apply Help

8 Verification Steps

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

8.1 Verify Avaya Aura® Communication Manager

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

status trunk 60/1	Page 1 of 3
TRUNK STATUS	
Trunk Group/Member: 0060/001	Service State: in-service/active
Port: T00044	Maintenance Busy? no
Signaling Group ID: 60	
IGAR Connection? no	
Connected Ports: T00134	

Page 2 of the **status trunk** command indicates the codec being used for the call and whether the call is shuffled. If the call is shuffled, the **Audio Connection Type** field would be set to *ip-direct*; otherwise, it would be set to *ip-tdm* as shown below.

status trunk 60/1	Page 2 of 3
CALL CONTROL SIGNALING	
Near-end Signaling Loc: PROCR	
Signaling IP Address	Port
Near-end: 192.168.100.10	: 5060
Far-end: 192.168.100.235	: 5060
H.245 Near:	
H.245 Far:	
H.245 Signaling Loc:	H.245 Tunneled in Q.931? no
Audio Connection Type: ip-tdm	Authentication Type: None
Near-end Audio Loc:	Codec Type: G.711MU
Audio IP Address	Port
Near-end: 192.168.100.15	: 2112
Far-end: 192.168.100.170	: 10002
Video Near:	
Video Far:	
Video Port:	
Video Near-end Codec:	Video Far-end Codec:

8.2 Verify Avaya Aura® Session Manager

Verify that the Hammer SIP trunks are up by navigating to **Home→Elements→Session Manager→System Status→SIP Entity Monitoring** and clicking on the appropriate SIP entities.

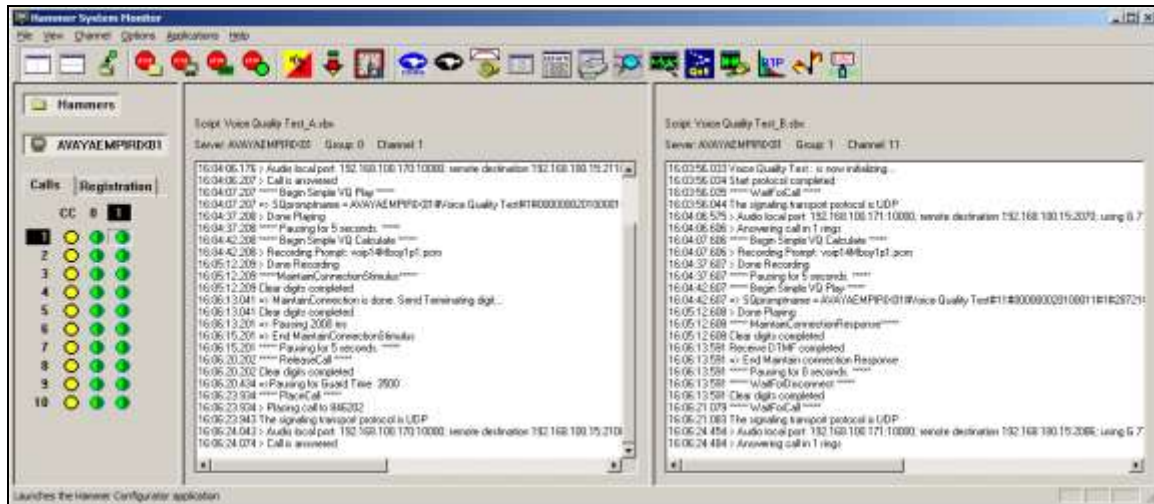
Below is the status of the SIP trunks used for incoming/outgoing calls from/to Hammer IP.

The screenshot displays the 'Session Manager Entity Link Connection Status' page. The left sidebar contains a navigation menu with options like Session Manager, Dashboard, Session Manager Administration, Communication Profile Editor, Network Configuration, Device and Location Configuration, Application Configuration, and System Status. The main content area shows the 'SIP Entity Monitoring' section. It includes a breadcrumb trail: Home / Elements / Session Manager / System Status / SIP Entity Monitoring. Below the title, there is a description: 'This page displays detailed connection status for all entity links from a Session Manager.' A link 'All Entity Links for Session Managers tz-aam' is present. A 'Summary view' button is also visible. A table with 8 columns (SIP Entity Name, SIP Entity Resolved IP, Port, Proto, Descr, Conn. Status, Reason Code, Link Status) displays 3 items. The first three items are 'Hammer-Inc', 'devcon14', and 'Hammer-Out', all with a Link Status of 'UP'.

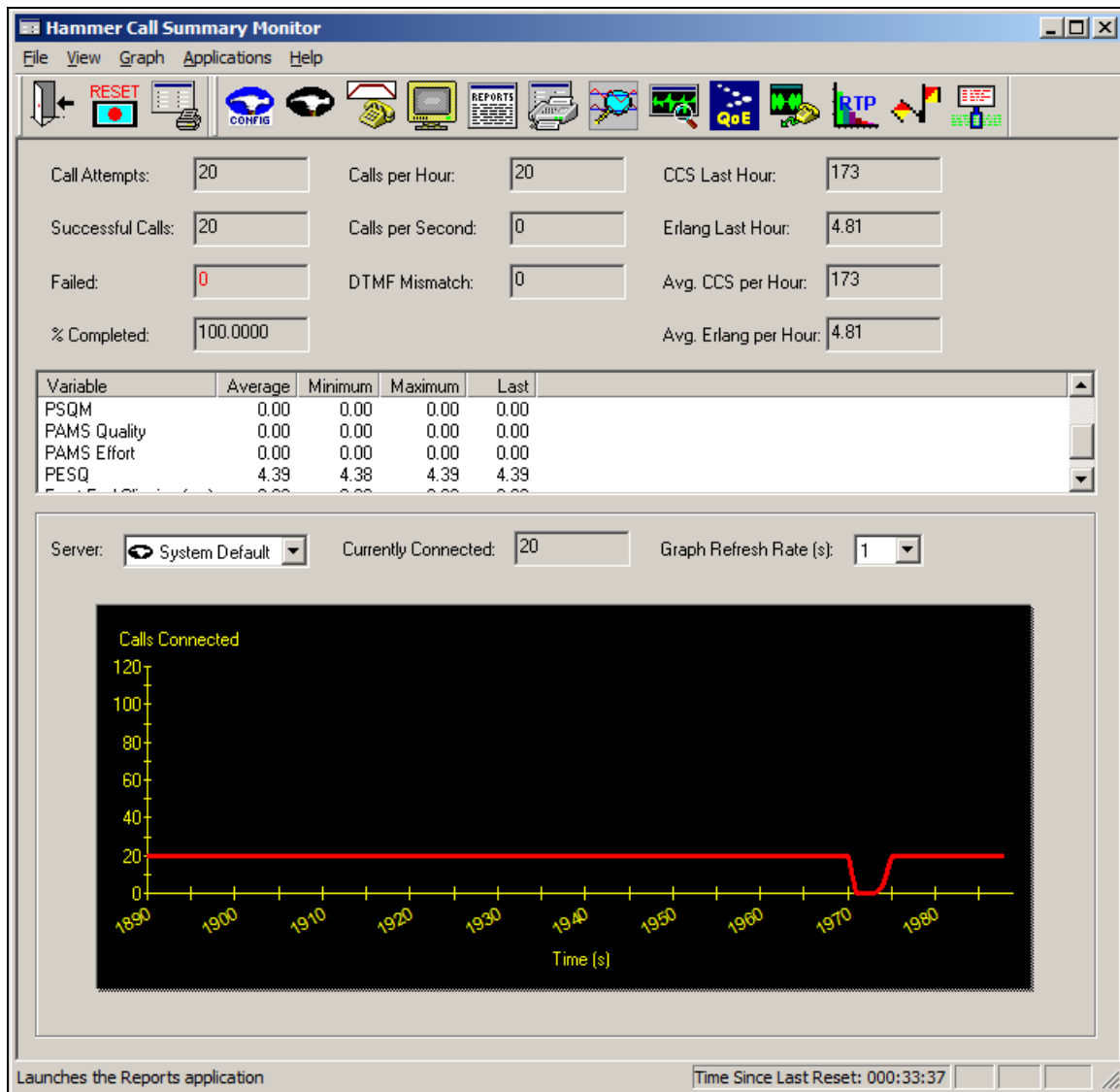
SIP Entity Name	SIP Entity Resolved IP	Port	Proto	Descr	Conn. Status	Reason Code	Link Status
Hammer-Inc	192.168.100.170	5060	UDP	FALSE	UP	200 OK	UP
devcon14	192.168.100.10	5060	TCP	FALSE	UP	200 OK	UP
Hammer-Out	192.168.100.171	5060	UDP	FALSE	UP	200 OK	UP

8.3 Verify Empirix Hammer IP

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



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.



9 Conclusion

These Application Notes describe the configuration steps required to integrate the Empirix Hammer IP with an Avaya SIP telephony network using SIP trunk emulation. The Hammer IP was able to establish a SIP trunk with Avaya Aura® Session Manager, exchange SIP Options messages, successfully establish calls through the Avaya SIP telephony network, generate voice quality metrics, monitor the calls, and generate reports. All feature and serviceability test cases were completed successfully.

10 References

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

- [1] *Administering Avaya Aura® Communication Manager*, Release 6.3, Issue 10.0, June 2014, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, Release 6.3, Issue 7, September 2014.
- [3] *Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager 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 using H.323 Endpoint Emulation*, Issue 1.0, available at <http://www.avaya.com>.
- [5] *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>.
- [6] *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 screenshot shows the 'Media Profile Editor' window with the title bar path: \\AVAYAEMPIRX01\Hammer\IPSigServer\SDPs\G711U.sdp. The window is divided into three main sections:

- Session Description:** A table with 'Include Field?' checkboxes and 'Field'/'Value' columns.

Include Field?	Field	Value
<input type="checkbox"/>	(o=) Owner:	Empirix VQ Agent
<input type="checkbox"/>	(s=) Session Name:	Empirix VQ Test Session
<input type="checkbox"/>	(i=) Session Information:	
<input type="checkbox"/>	(u=) URI of Description:	
<input type="checkbox"/>	(e=) Email Address:	
<input type="checkbox"/>	(p=) Phone Number:	
<input type="checkbox"/>	(b=) Bandwidth Information:	
- (a=) Attributes:** A large empty text area with 'Add', 'Edit', and 'Delete' buttons to its right.
- Media Descriptions:** Three buttons: 'Audio Description' (checked), 'Image (T.38) Description', and 'Video Description'.

At the bottom are buttons for 'New', 'Save', 'Load', 'Delete', 'Preview', 'OK', 'Cancel', and 'Help'.

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.

MPE Audio Description: \\AVAYAEMPIRX01\Hammer\IPSigServer\SDPs\G711U.sdp

Order and configure codecs to advertise in Media Profile

Codec	Send 'rtpmap'?	Payload Type
<input checked="" type="checkbox"/> G.711U	No	0
<input type="checkbox"/> G.711A	No	8
<input type="checkbox"/> G.723	No	4
<input type="checkbox"/> G.729A	No	18
<input type="checkbox"/> G.729AB	No	18
<input type="checkbox"/> G.726 40 kb/s	Yes	127
<input type="checkbox"/> G.726 32 kb/s	Yes	97
<input type="checkbox"/> G.726 24 kb/s	Yes	98
<input type="checkbox"/> G.726 16 kb/s	Yes	99
<input checked="" type="checkbox"/> RFC 2833	Yes	101
<input type="checkbox"/> G.728	Yes	122

Optional Descriptions

Include Field? Field Value

☐ (i=) Media Information:

☐ (b=) Bandwidth Information:

(a=) Attributes

Add
Edit
Delete

OK Cancel Help

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