



Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Trunk Emulation with Secure SIP (SIPS) / Transport Layer Security (TLS) – 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 with SIP trunk emulation using Secure SIP (SIPS). Hammer IP validates IP-based systems by testing the actual network under anticipated traffic conditions to provide a complete understanding of expected performance. Hammer IP can be used to assess and monitor network performance, reliability and quality of VoIP services in an Avaya IP telephony network. In this configuration, Hammer IP emulates SIP trunks that interface to Avaya Aura® Session Manager and originates and terminates calls through Avaya SIP telephony network. In addition, this solution supports SIPS to secure the SIP signaling using TLS (Transport Layer Security) and Secure Real-time Transport Protocol (SRTP) to protect the RTP data. 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 with Secure SIP (SIPS). Hammer IP validates IP-based systems by testing the actual network under anticipated traffic conditions to provide a complete understanding of expected performance. Hammer IP can be used to assess and monitor network performance, reliability and quality of VoIP services in an Avaya IP telephony network. In this configuration, Hammer IP emulates SIP trunks that interface to Avaya Aura® Session Manager and originates and terminates calls through Avaya SIP telephony network. In addition, this solution supports SIPS to secure the SIP signaling using TLS (Transport Layer Security) and Secure Real-time Transport Protocol (SRTP) to protect the RTP data. 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.

The following Application Notes are related to this solution.

- *Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Endpoint Emulation with Secure SIP (SIPS) / Transport Layer Security (TLS) [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 SIP trunk channels on Hammer IP and establishing the calls through the Avaya SIP telephony network using SIPS/TLS and SRTP. 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 interface to Avaya Aura® Session Manager as SIP trunks, establish calls, send voice media, and provide voice quality metrics using SIPS/TLS and SRTP. The following features and functionality were covered:

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

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

2.2 Test Results

All test cases passed. Empirix Hammer IP was successful in originating calls using SIP trunk emulation and terminating calls on channels emulating SIP endpoints and SIP trunks using SIPS/TLS and SRTP. The following observations were noted:

- This solution does not currently support Direct IP-IP Media (i.e., Shuffling) with SIP trunks using SIPS/TLS.
- The dial pattern used to route calls to the terminating Hammer SIP trunks should not match the **Trunk ANI** of the terminating Hammer SIP trunks. This is required so that the calls are routed to the appropriate terminating Hammer channel; otherwise, all calls get routed to the first terminating Hammer channel. See **Section 6.6** for more details.

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 using SIPS/TLS and SRTP. That is, the goal was to verify that Hammer IP can establish SIP trunks 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.

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 shown in **Figure 1** illustrates the test configuration. In this configuration, Session Manager receives calls from 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. SIP signaling is protected using SIPS/TLS and RTP data is protected using SRTP. 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.

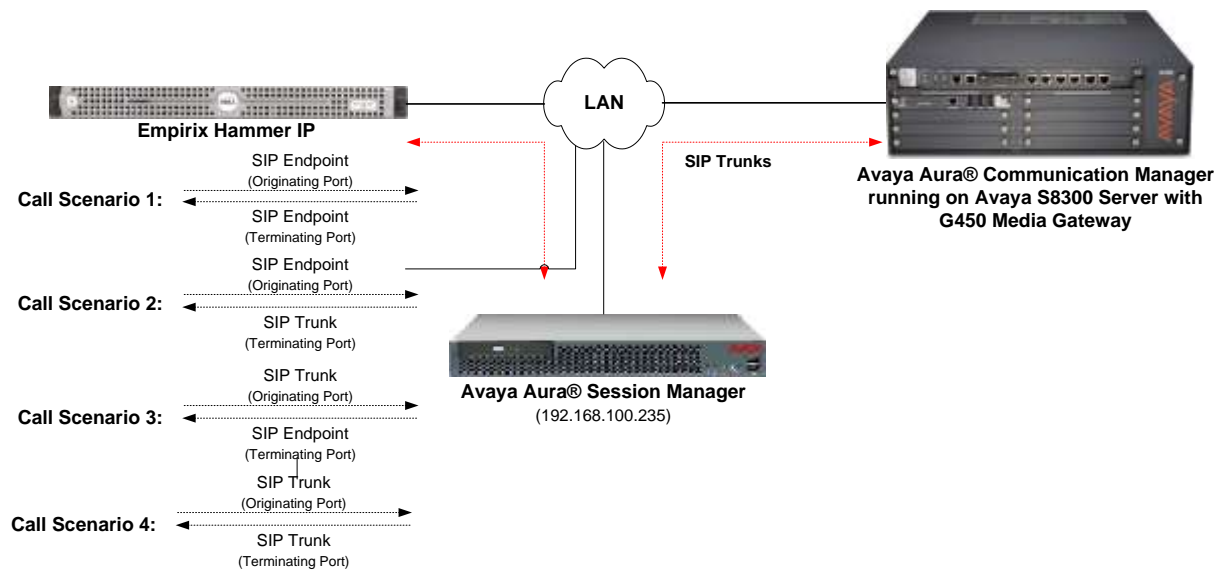


Figure 1: Empirix Hammer IP with Avaya SIP Telephony Network

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 SP 12 (R016x.03.0.124.0 with Patch 22505)
Avaya G450 Media Gateway	FW 36.12.0
Avaya Aura® System Manager	6.3.15 (Build No. 6.3.0.8.5682-6.3.8.5506 Software Update Revision No: 6.3.15.12.3972)
Avaya Aura® Session Manager running on an S8800 Server	6.3.15 (6.3.15.0.631503)
Empirix Hammer IP running on Microsoft Windows Server 2008 R2 with Dual 2.40 GHz Intel Xeon CPU and 12.0 GB of RAM	6.2.0.79

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 SIP Stations
- 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. If SRTP is required for the test, set **Media Encryption** to *1-srtp-aescm128-hmac80* as shown below. This is the media encryption supported by Hammer IP. Below is the IP codec set configured for G.711 mu-law and SRTP.

IP Codec Set

Codec Set: 1

	Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1:	G.711MU	n	2	20
2:				
3:				
4:				
5:				
6:				
7:				

Media Encryption

1: none
2: 1-srtp-aescm128-hmac80
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*.

IP NETWORK REGION

Region: 1
Location: 1 **Authoritative Domain: devcon.com**
Name:
MEDIA PARAMETERS **Intra-region IP-IP Direct Audio: no**
 Inter-region IP-IP Direct Audio: no
 IP Audio Hairpinning? n
 UDP Port Min: 2048
 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 *tls*. See **Section 7** for instructions on managing TLS certificates.
- The **Enforce SIPS URI for SRTP** field may be enabled if SIPS should be enforced when SRTP is being used.
- 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 *5061* 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 disabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.

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: tls	
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: 5061	Far-end Listen Port: 5061	
	Far-end Network Region: 1	
Far-end Domain: devcon.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? n	
Enable Layer 3 Test? y	IP Audio Hairpinning? 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 Administer 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.

```

change aar analysis 4
                                     Page 1 of 2

                                AAR DIGIT ANALYSIS TABLE
                                Location: all                Percent Full: 2

```

	Dialed String	Total Min Max	Route Pattern	Call Type	Node Num	ANI 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

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 No	NPA Mrk	Pfx Lmt	Hop List	Toll Del	No. Digits	Inserted Dgts	DCS/ QSIG Intw	IXC user
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.



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 Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

The screenshot shows the 'Location Details' form in the Avaya Aura System Manager 6.3. The 'General' tab is selected, and the 'Name' field is filled with 'Lincroft'. The 'Notes' field contains 'DevConnect Network'. Below this, there are sections for 'Dial Plan Transparency in Survivable Mode' and 'Overall Managed Bandwidth'. The 'Managed Bandwidth Units' are set to 'Kbit/sec'. The 'Total Bandwidth' and 'Multimedia Bandwidth' fields are empty. The 'Audio Calls Can Take Multimedia Bandwidth' checkbox is checked. The left sidebar shows the navigation menu with 'Locations' selected.

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.

The screenshot shows the 'Location Pattern' form. It has a table with one item: 'IP Address Pattern'. The pattern is '192.168.100.*' and the notes are 'devcon14 (CM) & lz-asm (SM)'. The 'Filter' is set to 'Enable'. The 'Commit' button is visible at the bottom.

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 login status 'Last Logged in at November 3, 2015 3:56 PM' with a 'Log off admin' link. A left-hand menu lists various configuration categories: 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 'SIP Entity Details' and 'General'. It contains several input fields: 'Name' (with value '12-asm'), 'FQDN or IP Address' (with value '192.168.100.235'), 'Type' (a dropdown menu set to 'Session Manager'), 'Notes' (an empty text area), 'Location' (a dropdown menu set to 'Lincroft'), 'Outbound Proxy' (an empty dropdown menu), 'Time Zone' (a dropdown menu set to 'America/New_York'), and 'Credential name' (an empty text field). At the bottom, there is a 'SIP Link Monitoring' section with 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 configuration window titled "Port". At the top, there are two input fields: "TCP Failover port:" and "TLS Failover port:", each followed by a text box. Below these are two buttons: "Add" and "Remove".

Below the buttons is a table with the following structure:

Port	Protocol	Default Domain	Notes
5060	TCP	devcon.com	
5060	UDP	devcon.com	
5061	TLS	devcon.com	

At the bottom of the window, there is a "Select" dropdown menu with options "All" and "None".

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 left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities (selected), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'SIP Entity Details' and shows the 'General' tab. The form includes the following fields: Name (devcon14), FQDN or IP Address (192.168.100.10), Type (CM), Notes, Adaptation, Location (Lincoln), Time Zone (America/New_York), SIP Timer B/F (in seconds) (4), Credential name, and Call Detail Recording (none). Buttons for 'Commit' and 'Cancel' are visible at the top right of the form area. The top of the interface shows the user is logged in as 'admin' on November 3, 2015, at 3:34 PM.

6.3.3 Empirix Hammer IP

Two SIP Entities must be added for Empirix Hammer IP, one for incoming calls to Session Manager and another one for outgoing calls to Hammer IP. 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 sidebar shows a navigation menu with options like Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The 'SIP Entities' option is selected. 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:** Lincoln
- Time Zone:** America/New_York
- SIP Timer R/F (in seconds):** 4
- Credential name:** (empty)
- Call Detail Recording:** egress
- Loop Detection Node:** 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 November 23, 2016, at 1:03 PM.

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 'Log off admin' button. The left sidebar contains a menu with options: Routing, Domains, Locations, Adaptations, SIP Entities (selected), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area shows the 'SIP Entity Details' page for 'Hammer-Out'. The breadcrumb trail is 'Home / Elements / Routing / SIP Entities'. The 'General' tab is active, showing fields for Name (Hammer-Out), FQDN or IP Address (192.168.100.171), Type (SIP Trunk), Notes, Adaptation, Location (Lincroft), 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 at the top right of the form.

AVAYA
Aura System Manager 6.3

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: Lincroft

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

6.4 Add Entity Links

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

Note: See **Section 7** for instructions on managing TLS certificates.

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 (e.g., *TLS*).
- **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.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DRS Override	Port	Connection Policy	Delay New Service
* lz-asm to devcon14 L	* lz-asm	* TLS	* 3061	* devcon14	<input type="checkbox"/>	* 3061	* trusted	<input type="checkbox"/>

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 (e.g., *TLS*).
- **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*).



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 6.3 interface. The left sidebar shows the navigation menu with 'Routing Policies' selected. The main content area is titled 'Name / Elements / Routing / Routing Policies'. The 'Routing Policy Details' section is active, showing the 'General' tab. The 'Name' field is set to 'devcon14 Policy'. The 'Disabled' checkbox is unchecked. The 'Retries' field is set to 0. The 'Notes' field is empty. Below the 'General' tab is the 'SIP Entity as Destination' section, which includes a 'Select' button and a table with the following data:

Name	FQDN or IP Address	Type	Notes
devcon14	192.168.105.10	CM	

Below the table is the 'Time of Day' section, which includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. A table shows the time range configuration:

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

The 'Filter: Enable' button is visible in the top right corner of the Time of Day section.

The following screen shows the Routing Policy for 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 indicator 'Last logged on at November 23, 2015 1:12 PM' with a 'Log off admin' link. A left-hand menu lists various configuration areas: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies (highlighted), 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 'Continue' and 'Cancel' buttons. Below this is the 'General' tab, which contains fields for 'Name' (set to 'Hammer Policy'), a 'Disabled' checkbox, 'Retries' (set to 0), and a 'Notes' field. The 'SIP Entity as Destination' section includes a 'Select' button and a table with columns 'Name', 'FQDN or IP Address', 'Type', and 'Notes'. The table lists 'Hammer-Out' with IP '192.168.100.171' and type 'SIP Trunk'. The 'Time of Day' section has 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. It shows '1 Item' with a 'Filter: Enable' option. A table below lists the time range '24/7' with start and end times of '00:00' and '23:59'. The bottom of the interface shows a 'Select: All, None' option.

Avaya
Aura System Manager 6.3

Last logged on at November 23, 2015 1:12 PM
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 / Routing Policies

Routing Policy Details
Continue Cancel

General

* Name: Hammer Policy

Disabled: ☐

* Retries: 0

Notes:

SIP Entity 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 Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7								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 "46200" will be routed to Communication Manager. The '8' is the AAR access code and "46200" are the digits routed to the Hammer IP, which will terminate on a SIP trunk.

Note: The dial pattern should not match the **Trunk ANI** of the terminating Hammer SIP trunks configured in **Section 8.2.2.1**. For example, the **Trunk ANI** of the terminating Hammer SIP trunks was 46202. A different dial pattern should be used instead of 46202.

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.

Avaya Aura System Manager 6.0

Last logged on at November 23, 2015 1:23 PM

Home / Elements / Routing / Dial Patterns

Dial Pattern Details [Commit] [Cancel] [Help ?]

General

* Pattern: 846

* Min: 6

* Max: 6

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
Unicast	DevConnect Network	devcon14 Policy	0	<input type="checkbox"/>	devcon14	

Select: All, None

The following screen shows the dial pattern definition for Hammer IP. The extension, 46200, will be routed to Hammer IP. Note that “46200” 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 [Commit] [Cancel] [Help ?]

General

* Pattern: 462

* Min: 5

* Max: 5

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: devconn.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
Lincoft	DevConned Network	Hammer Policy	0	<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.

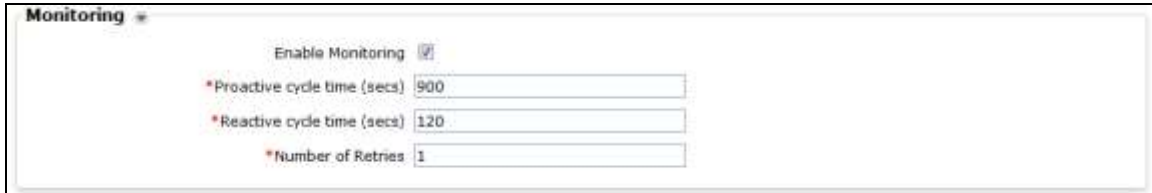
Under *Security Module*:

- **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 displays the 'Edit Session Manager' configuration interface in Avaya Aura System Manager 6.3. 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', 'System States', 'System Tools', and 'Performance'. The main content area is titled 'Edit Session Manager' and includes 'Commit' and 'Cancel' buttons. Below the title, there are tabs for 'General' and 'Security Module'. The 'General' tab is active, showing fields for 'SIP Entity Name' (Iz-asm), 'Description', and 'Management Access Point Host Name/IP' (192.168.100.233). The 'Security Module' tab is also visible, showing fields for 'SIP Entity IP Address' (192.168.100.235), '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 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 Managing and Creating TLS Certificates

This section covers how to manage and create the TLS certificates required to support TLS over SIP trunks between Communication Manager and Session Manager and to support TLS for the emulated SIP endpoints in Hammer IP. For this solution, Avaya Aura® System Manager is used as a certificate authority. For additional information on managing TLS certificates, refer to [2]. The steps are required include:

- Export the System Manager CA Certificate
- Add the System Manager CA to Communication Manager
- Install Enhanced Validation Certificates for Session Manager
- Create TLS Certificate and Private Key for Hammer IP

7.1 Export the System Manager CA Certificate

To export the System Manager CA, follow these steps:

1. On the home page of the System Manager Web Console, under **Services**, select **Security**→**Certificates**→**Authority**.
2. On the CA Functions page, click **Download pem file**.



3. Save the file. In this example, the filename was SMGR_CA.pem.

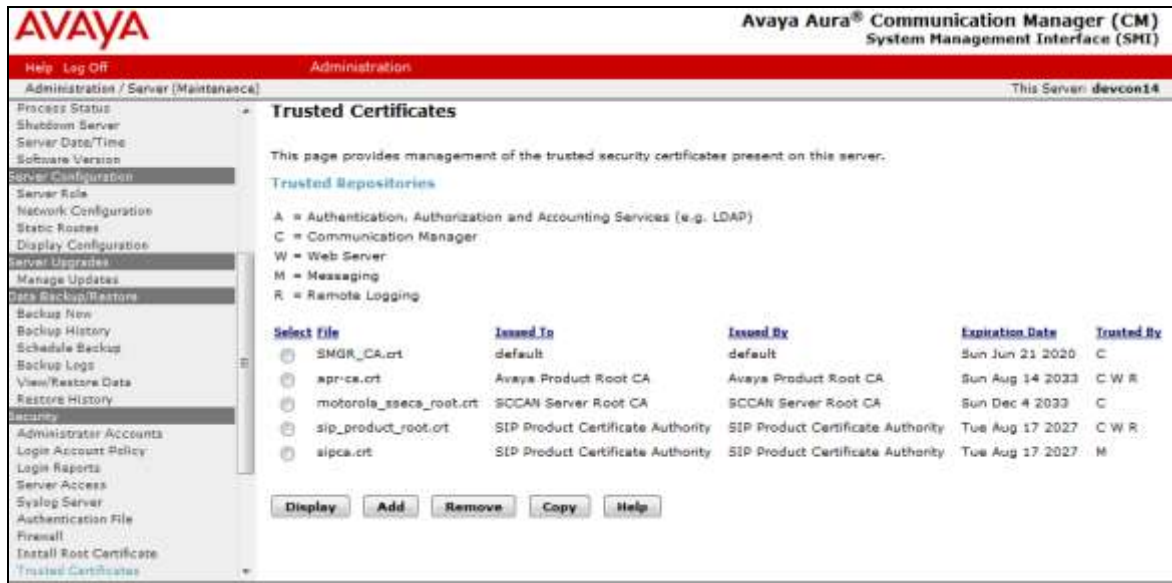
7.2 Add the System Manager CA to Communication Manager

Use the following procedure to make Communication Manager trust the System Manager CA certificate.

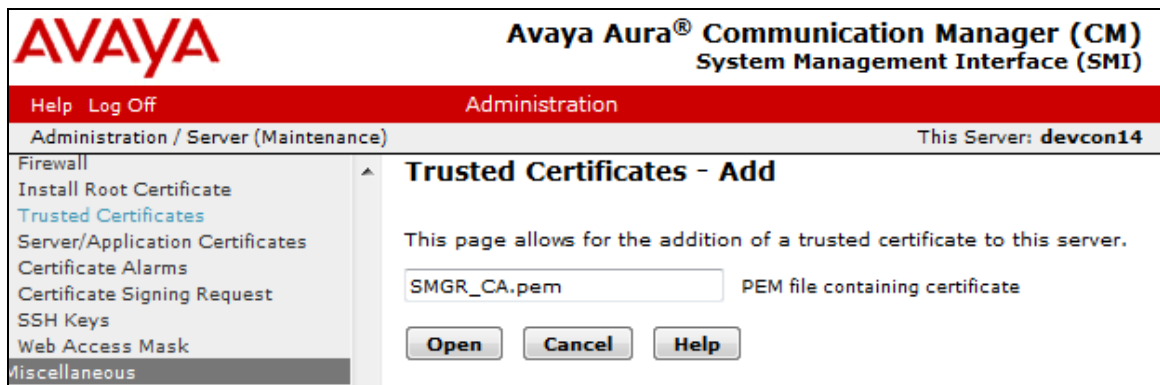
1. Verify the System Manager CA certificate downloaded in **Section 7.1** can be accessed.
2. Log into the Communication Manager server web interface.
3. Click **Administration** and select **Service (Maintenance)**.
4. In the left pane, under **Miscellaneous**, click **Download Files**.
5. Select **File(s) to download from the machine I'm using to connect to the server** and click **Browse**.
6. Select the System Manager CA certificate to download as shown below and click **Download**.

The screenshot shows the Avaya Aura Communication Manager (CM) System Management Interface (SMI) Administration page. The left sidebar contains a navigation menu with categories: Process Status, Server Configuration, Server Upgrades, Data Backup/Restore, Security, and Miscellaneous. The 'Download Files' option is selected under the Miscellaneous category. The main content area is titled 'Download Files' and contains the following text: 'The Download Files SMI page lets you download files to the server.' Below this text are two radio button options. The first option, 'File(s) to download from the machine I'm using to connect to the server', is selected. It has a 'Browse...' button next to it, which has been clicked, showing 'SMGR_CA.pem' in the text field. Below this are three more 'Browse...' buttons, each with 'No file selected.' next to it. The second option, 'File(s) to download from the LAN using URL', is unselected and has three empty text input fields below it. At the bottom, there is a 'Proxy Server' text input field with the placeholder '(e.g proxy.domain:3152)'. Below the proxy server field are 'Download' and 'Help' buttons. The top of the page has a red header with the Avaya logo and the text 'Avaya Aura® Communication Manager (CM) System Management Interface (SMI)'. Below the header is a navigation bar with 'Help Log Off' and 'Administration' links. The current page is 'Administration / Server (Maintenance)' and the server name is 'devcon14'.

7. In the left pane, under **Security**, click **Trusted Certificates** and click **Add**.



8. Enter the name of the downloaded System Manager CA certificate as shown below and click **Open**.



9. Enter the file name again in the text box and select the Communication Manager checkbox as shown below and click **Add**.

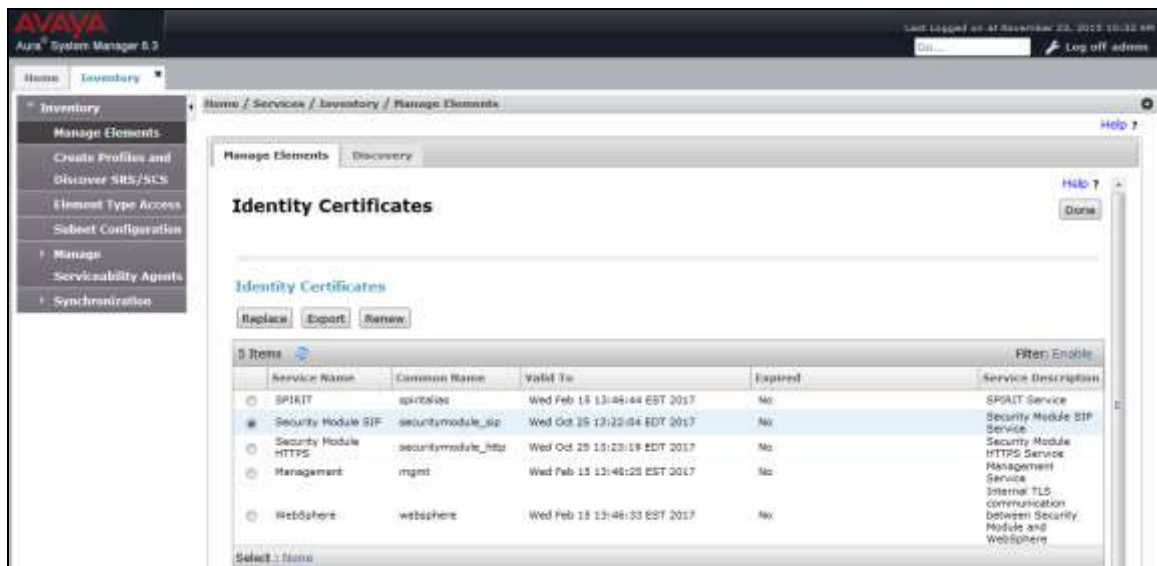
The screenshot displays the Avaya Aura Communication Manager (CM) System Management Interface (SMI). The top navigation bar includes 'Help', 'Log Off', and 'Administration'. Below this, a red banner shows 'Administration / Server (Maintenance)' and 'This Server: devcon14'. A left-hand menu lists various system management tasks, with 'Trusted Certificates' highlighted under the 'Security' section. The main content area, titled 'Trusted Certificates', explains that this page manages trusted security certificates. It features a table with columns 'Issued To', 'Issued By', and 'Expiration Date', showing a single entry: 'default', 'default', and 'Sun Jun 21 2020'. Below the table is a text input field containing 'SMGR_CA.pem' and a label 'Store the certificate in this file in each repository selected below'. A section titled 'Add to these trusted repositories' contains five checkboxes: 'Authentication, Authorization and Accounting Services (e.g. LDAP)', 'Communication Manager' (which is checked), 'Web Server', 'Messaging', and 'Remote Logging'. At the bottom of this section are three buttons: 'Add', 'Cancel', and 'Help'.

10. Restart Communication Manager.

7.3 Install Enhanced Validation Certificates for Session Manager

Perform this procedure to populate the **Common Name** and **Subject Alternate Name** of the certificate in Session Manager.

1. On the home page of the System Manager Web Console, under **Services**, click **Inventory**→**Manage Elements**.
2. Select the appropriate Session Manager Web Console from the list and click **More Actions**.
3. Select **Configure Identity Certificates** from the drop-down menu.
4. On the **Identity Certificates** page, select **Security Module SIP** and click **Replace**.



5. On the **Replace Identity Certificate** page, select **Replace this Certificate with Internal CA Signed Certificate**.
6. Select the **Common Name (CN)** checkbox and enter the host name or IP address of the Security Module. The address is the same as the SIP Entity address.
7. Select **RSA** for the **Key Algorithm**.
8. Select **2048** as the **Key Size**.

9. Select the **DNS Name** checkbox and enter the SIP domain (e.g., *devcon.com*) and click **Commit**.

The screenshot shows the 'Replace Identity Certificate' page in the Avaya Aura System Manager 8.3 interface. The page is divided into a left sidebar with navigation links and a main content area. The main content area has a title bar 'Replace Identity Certificate' and a 'Help' link. Below the title bar, there are two tabs: 'Manage Elements' and 'Discovery'. The 'Manage Elements' tab is active. The page contains several input fields and checkboxes for configuring the certificate replacement. The 'Subject Details' section includes fields for 'Subject Details', 'Valid From', 'Valid To', 'Key Size', 'Issuer Name', 'Certificate Fingerprint', and 'Subject Alternative Name'. The 'Subject Alternative Name' field has the 'DNS Name' checkbox selected and the value 'devcon.com' entered. Below the 'Subject Details' section, there are two radio buttons: 'Replace this Certificate with Internal CA Signed Certificate' (selected) and 'Import third party certificate'. Further down, there are checkboxes for 'Common Name' and 'Key Algorithm' (RSA), and a 'Key Size' field set to 2048. At the bottom, there are checkboxes for 'Subject Alternative Name', 'DNS Name' (selected), 'IP Address', and 'URL'. The page has a left sidebar with navigation links and a top header with the Avaya logo and system information.

10. On the **Identity Certificate** page, select **Security Module HTTP** and click **Replace**.
The following steps are similar to the ones covered for **Security Module SIP** above.
11. On the **Replace Identity Certificate** page, select **Replace this Certificate with Internal CA Signed Certificate**.
12. Select the **Common Name (CN)** checkbox and enter the host name or IP address of the Security Module. The address is the same as the SIP Entity address.
13. Select **RSA** for the **Key Algorithm**.
14. Select **2048** as the **Key Size**.
15. Select the **DNS Name** checkbox and enter the SIP domain (e.g., *devcon.com*) and click **Commit**.
16. Click **Commit**.

7.4 Create TLS Certificate and Private Key for Hammer IP

This section covers the procedures for creating TLS Certificate and Private Key files for Hammer IP. Refer to [2] for more information on creating TLS Certificate and Private Key files.

7.4.1 Create Avaya Private Key Certificates

Follow the following procedure to add Hammer IP as an end entity.

1. Create **End Entity** for Hammer IP. From the System Manager home page, navigate to **Security→Certificates→Authority→RA Functions** and select **Add End Entity**.
2. Enter the following values and use the default values for the remaining fields. For the **CN, Common Name** field, any IP address associated with Hammer IP may be used. Click **Add End Entity** to submit.

3. Navigate to **Security→Certificates→Authority→Public Web**. The EJBCA window is displayed.
4. Click on **Create Keystore**.

- Under the **Authentication** section, enter the user name and password that were defined in Step 2, and then click OK.

EJBCA

EJBCA Certificate Enrollment

Welcome to certificate enrollment.

Please enter your username and password. Then click OK to generate your token.

Authentication

Username: empirix

Password: *****

OK

- In the **Options** section of the **EJBCA Token Certificate Enrollment** page, select *2048 bits* for the **Key length** field and click **OK** to continue.

EJBCA

EJBCA Token Certificate Enrollment

Welcome to certificate enrollment.

If you want to, you can manually install the CA certificate(s) in your browser, otherwise this will be done automatically when your certificate is retrieved.

Install CA certificates:

[Certificate chain](#)

Please choose a key length, then click OK to fetch your certificate.

Tick the "OpenVPN" checkbox if you want to create an OpenVPN installer. This options requires special configuration of the CA.

Options

Leave values as default if unsure.

Key length: 2048 bits

Certificate profile: ID_CLIENT_SERVER

OpenVPN installer: ☐

OK

- In the next window, click **Save** to save to the file to the local PC. This file will contain a Private Key, Server Certificate, and Trusted Root Certificate.

7.4.2 Create TLS Certificates and Private Key Files for Hammer IP

The following procedure describes how to create a TLS Certificate and TLS Private Key file for Hammer IP.

1. Using a Text Editor, open the Private Key Certificate file created in the previous section.

```
Bag Attributes
  friendlyName: 192.168.100.170
-----BEGIN PRIVATE KEY-----
MIIEvgIBADANBgkqhkiG9w0BAQEFAASCBAgEAAoIBAQDKDk9O2J/fwk/c
A7sPM+ffZ7JKQrjyXa8Yye3vQwJLUMptlMgmDP+LuJhDKicxBFpkqS2yhP7xGIO3
. . .
NIPZmEhDPRBQfGeY27pq1H+kcv+GsV+ogMn+r2IQ/tiYNkBhLGCRkq+f2VVQ+4R6
7zJI2WvYqRVjc/Hv8z6wH3mu+8aI3E/6Z6wmJuKJelvuWK0i/7gLZqhBFXqhCPHl
LRvylJQkVjBnZqoDeEFmr+o7
-----END PRIVATE KEY-----
Bag Attributes
  friendlyName: 192.168.100.170
subject=/CN=192.168.100.170/OU=SDP/O=AVAYA/C=US
issuer=/CN=default/OU=MGMT/O=AVAYA
-----BEGIN CERTIFICATE-----
MIIC9jCCAl+gAwIBAgIIL67fE/hqP0AwDQYJKoZIhvcNAQELBQAwMTEQMA4GA1UE
AwwHZGVmYXVsdDENMAsgAlUECwwETUdNVDEOMAwGA1UECgwFQVZBWUEwHhcNMTUx
. . .
+vIRd580oxvZvdltMjw1M17GPf6xNrGGnxPpjKQ29kRC1hfxQEjHVBfjXcwXy0vU
X8MSgDrrg8b4Mdv93OUMwvk62IzPmrrJfH3NuyjWoRVjqI35KVX6hIF6
-----END CERTIFICATE-----
Bag Attributes
  friendlyName: default
subject=/CN=default/OU=MGMT/O=AVAYA
issuer=/CN=default/OU=MGMT/O=AVAYA
-----BEGIN CERTIFICATE-----
MIICQjCCAaugAwIBAgIIDBlg8Zh4CWMwDQYJKoZIhvcNAQEFBQAwMTEQMA4GA1UE
AwwHZGVmYXVsdDENMAsgAlUECwwETUdNVDEOMAwGA1UECgwFQVZBWUEwHhcNMTAw
. . .
QrlRzEDot5Ep9d+SxWJOLptvmV6efH24ChSPyhyERDqcJNmi13vcEoINsW6GOBRx
V+OX1ILAXSJBBoxyakR00TL37pKnk8+lywRx9JjMqPzSj5TUljCT20BeI8MWk4LZP
D07sRnER
-----END CERTIFICATE-----
```

2. Copy the Private Key part of the file (i.e., the yellow highlighted lines starting with BEGIN PRIVATE KEY and ending with END PRIVATE KEY) and store it in a file. In this example, the file name was empirix_PrivateKey.pem.
3. Copy the Identity Certificate part of the file (i.e., the green highlighted lines starting with the first instance of BEGIN CERTIFICATE and ending with END CERTIFICATE) and store it in a file. In this example, the file name was empirix_Cert1.pem.
4. The empirix_PrivateKey.pem and empirix_Cert1.pem files must be used to configure the **TLS Certificate** and **TLS Private Key** of the originating and terminating

Hammer channels in the **Signaling** tab configured in **Sections 8.2.1** and **8.2.2.1**, respectively.

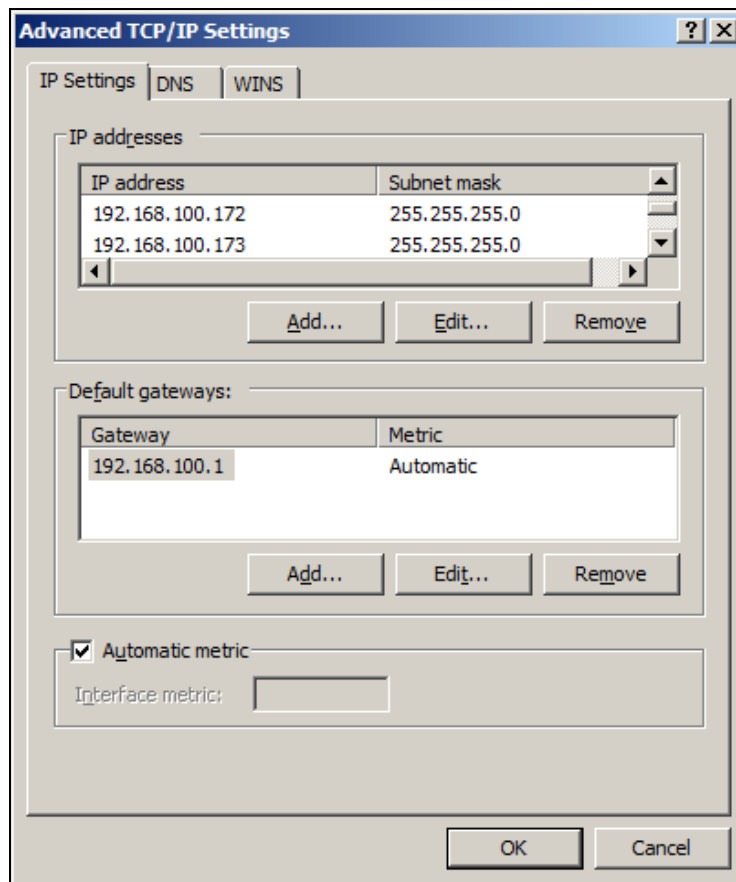
8 Configure Empirix Hammer IP

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, 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**.

8.1 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 (not shown) in Windows Server 2008.



8.2 Configure System

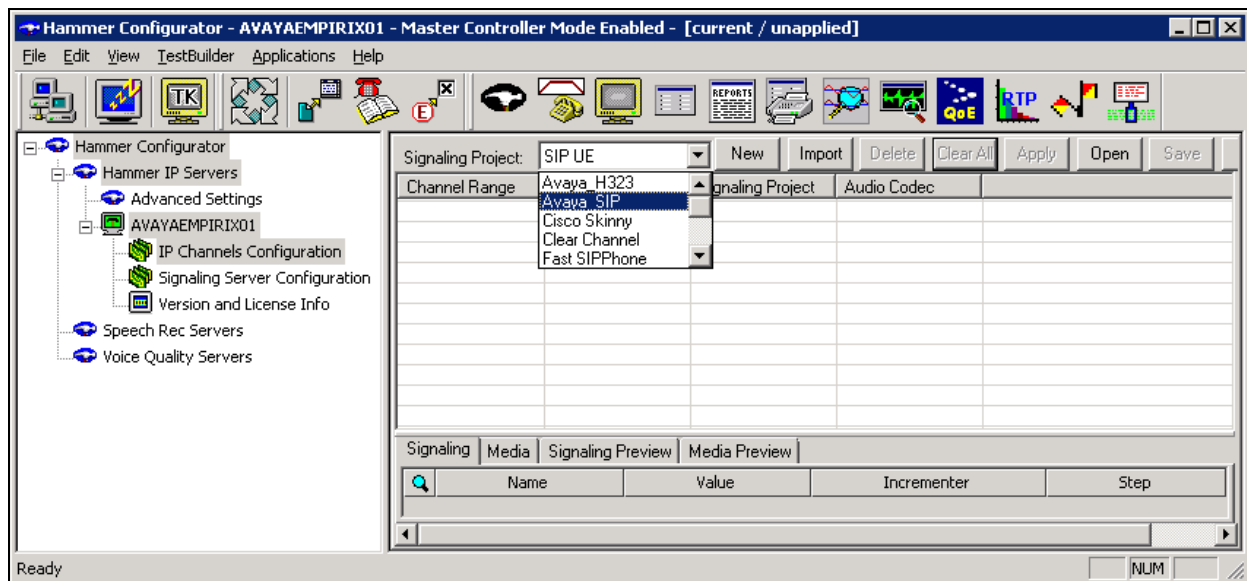
This section covers the configuration of originating and terminating channels on Hammer IP. In this configuration, the originating channels emulate SIP trunks. The terminating channels can emulate SIP endpoints or SIP trunks. These Application Notes will explicitly describe the configuration for terminating calls to SIP trunks in **Section 8.2.2.1**. In addition, it will provide a reference to other Application Notes for configuring terminating channels as SIP endpoints in **Section 8.2.2.2**.

8.2.1 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.

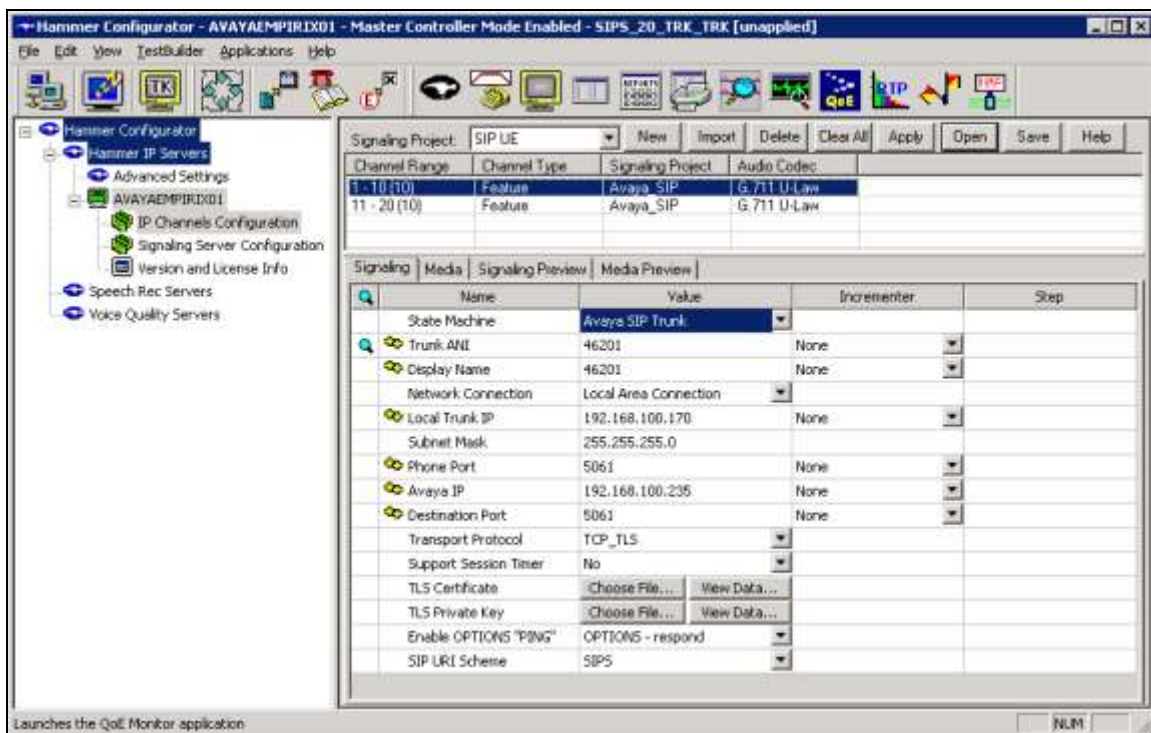
Note: It is assumed that Hammer IP is already in **Master Controller Mode**. To verify, check that the title bar of the **Hammer Configurator** indicates *Master Controller Mode Enabled* as shown below. It is also assumed that a system was already added to the configuration. In this configuration, the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



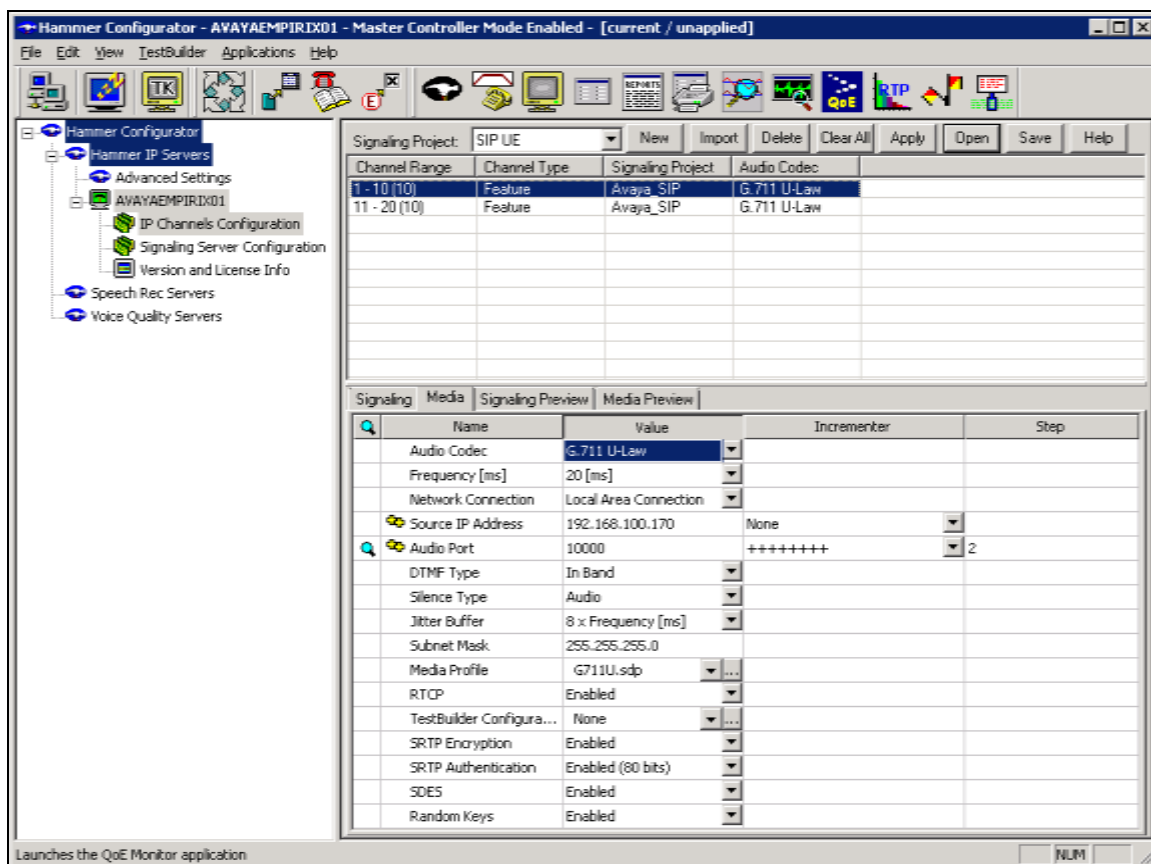
The first line in the grid that is highlighted in the figure below corresponds to the 10 originating channels. To set the number of channels in the group, click on the **Channel Range** cell in the grid and enter the number *10*. The following fields in the **Signaling** tab should be set as follows:

- **State Machine** should be set to *Avaya SIP Trunk*.
- **Trunk ANI** may be any extension but cannot match the dial pattern used to route calls to Hammer. See **Section 6.6**.
- **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**. 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*).
- **Phone Port** should be set to TLS port *5061*.
- **Avaya IP** should be set to the Session Manager SIP interface (e.g., *192.168.100.235*).
- **Destination Port** should be set to TLS port *5061*.
- **Transport Protocol** should be set to *TCP_TLS*. See **Section 7.4** for instructions on managing and creating TLS certificates.
- **Enable OPTIONS “PING”** should be set to *OPTIONS - respond*.
- **TLS Certificate** should be imported by clicking the **Choose File** button. Creating the TKS Certificate file is described in **Section 7.4**.
- **TLS Private Key** should be imported by clicking the **Choose File** button. Creating the TKS Private Key file is described in **Section 7.4**.
- **SIP URI Scheme** should be set to *SIPS*.
- 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**.
- **SRTP Encryption** should be enabled. Disable SRTP if not required for test.
- **SRTP Authentication** should be set to **Enabled (80 bits)**.
- **SDES** should be enabled.
- **Random Keys** should be enabled.
- The default values for the remaining fields may be used as shown.



8.2.2 Configure Terminating Channels

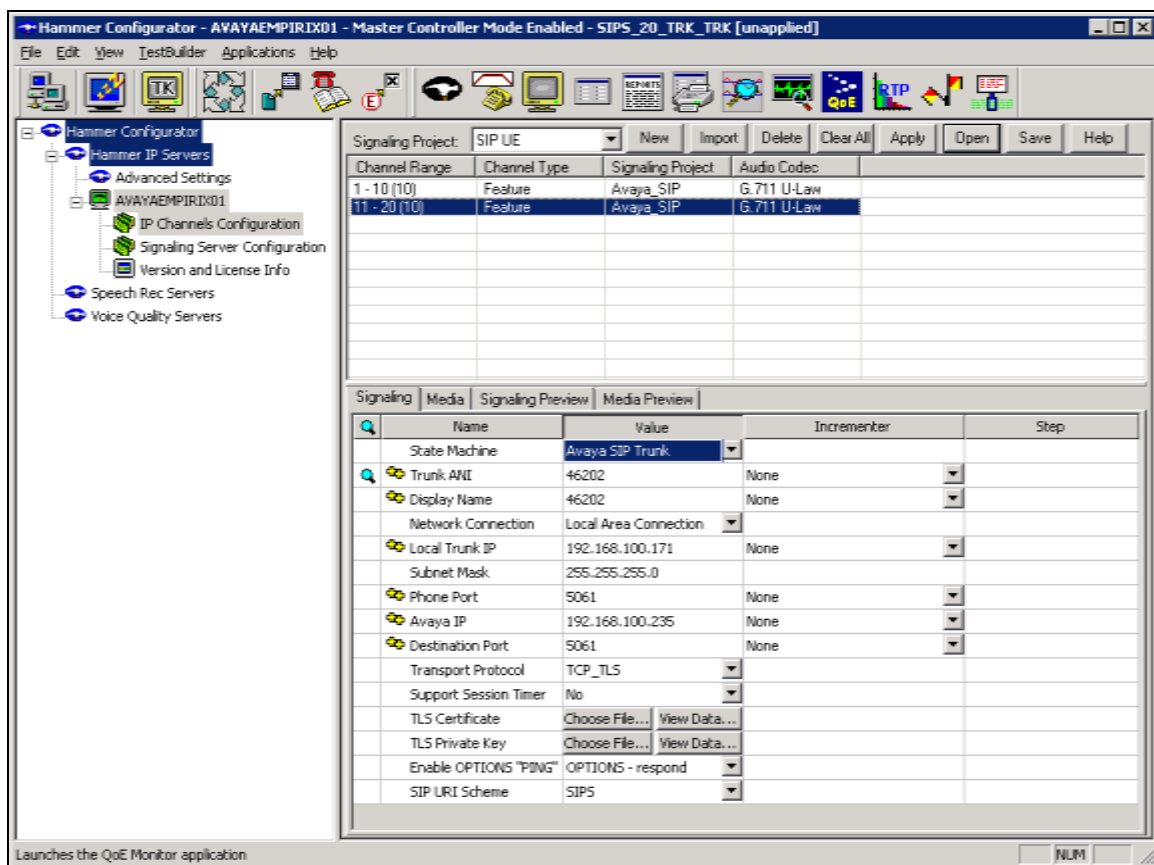
During the compliance test, the originating channels emulated SIP trunks with the calls terminating on SIP endpoints and SIP trunks. Select one of the following subsections depending on the configuration desired.

- **Section 8.2.2.1** for terminating calls on SIP trunks.
- **Section 8.2.2.2** for terminating calls on SIP endpoints.

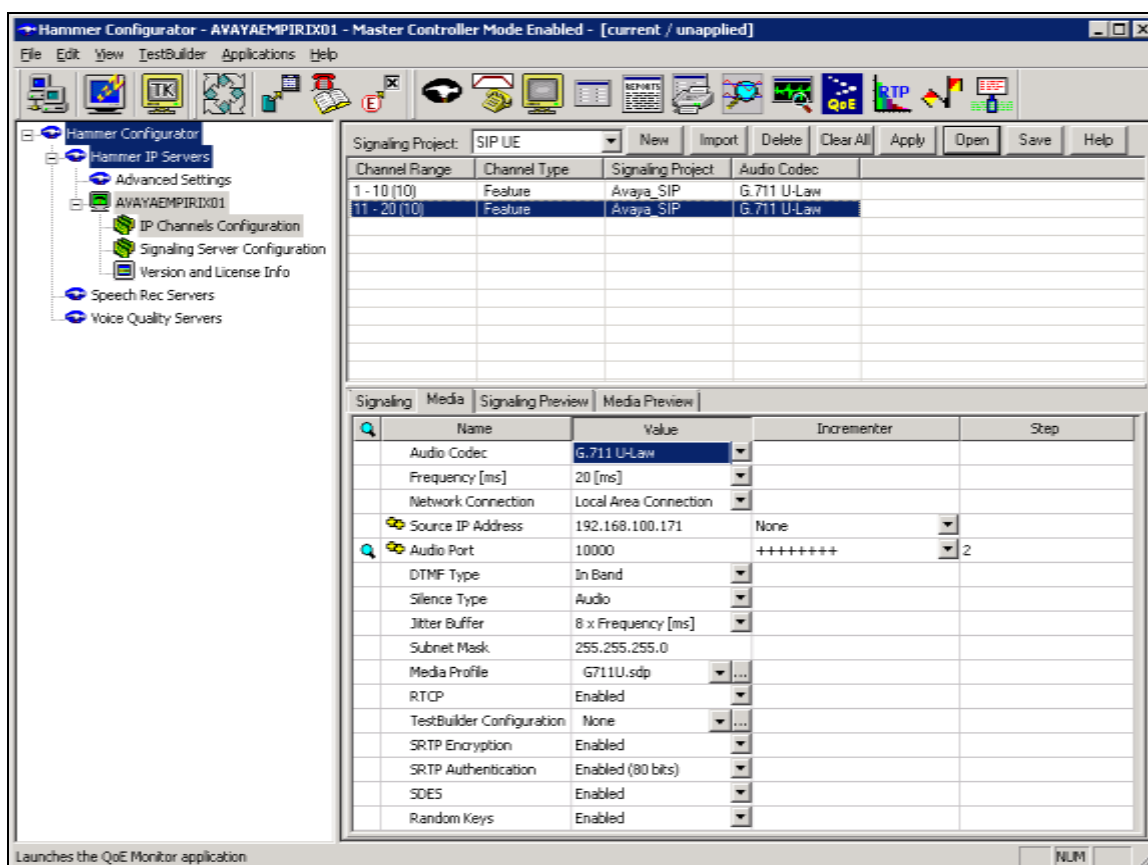
Note: Ensure that the originating and terminating channels are assigned unique IP addresses.

8.2.2.1 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 8.2.1** 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 addresses from *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.



8.2.2.2 Configure Terminating Channels – SIP Endpoints

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

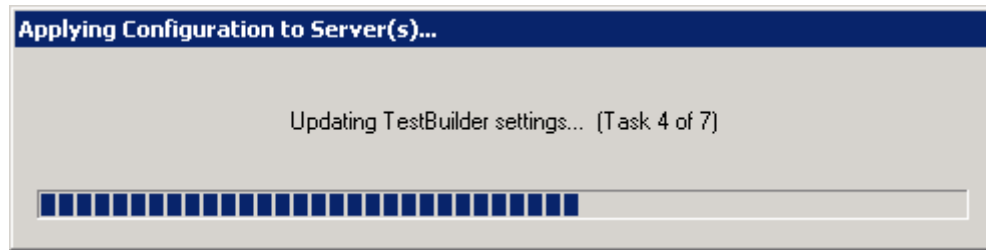
- **Section 5** describes how to configure SIP stations and call routing on Communication Manager.
- **Section 6** describes how to configure SIP endpoints on Session Manager.
- **Section 7.2.2.1** describes how to configure terminating SIP endpoints on Hammer IP.
- **Section 7.2.3** describes how to 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 [4] must be completed for terminating calls to SIP endpoints.

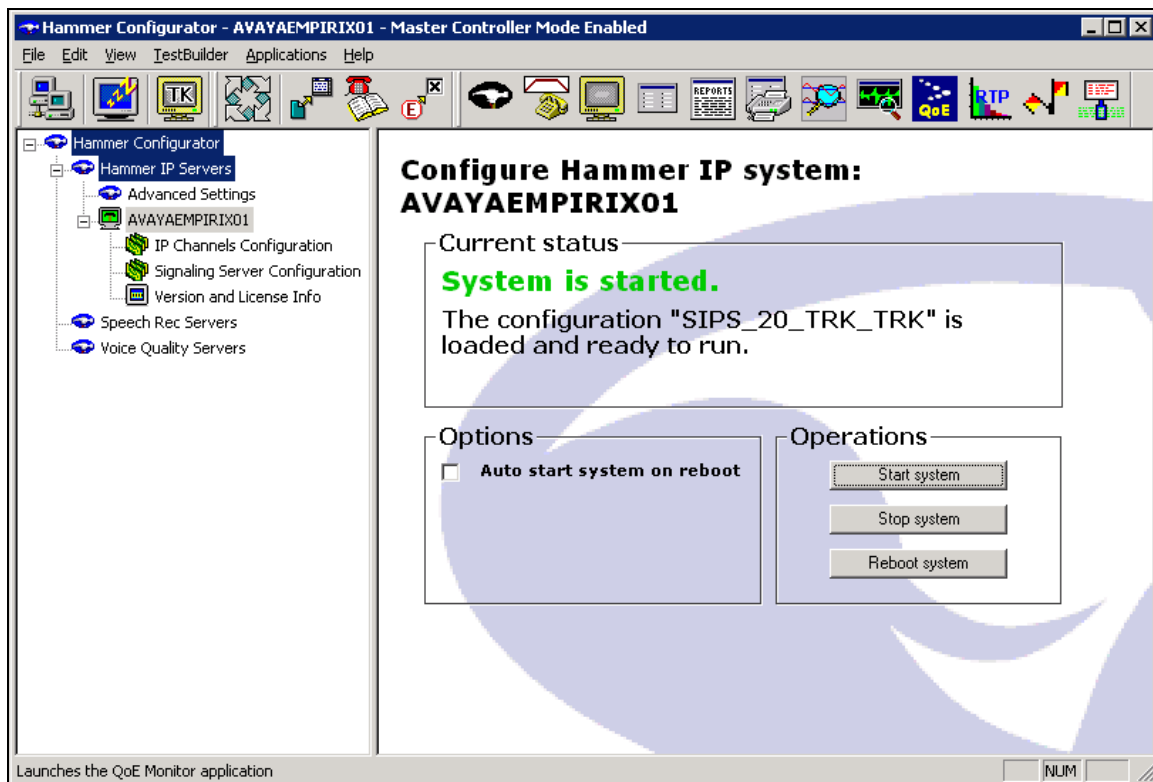
8.3 Applying the Hammer IP Configuration

This completes the configuration of Hammer IP. This configuration should be saved by clicking the **Save** button (not shown) on the **Hammer Configurator** window. The configuration needs to be applied to the server for the changes to take effect. Click on the **Apply** button (not shown) in

the **Hammer Configurator** window. The following window is displayed as the configuration is being applied to the server.



Check that the system has been started by clicking on the server name (e.g., *AVAYAEMPIRIX01*) in the left pane of the **Hammer Configurator**. If the current status is *System Is Stopped*, click the **Start system** button to start the system. When the system is started, it should appear as shown below and should also specify which configuration has been applied. The configuration performed above was saved as *SIPS_20_TRK_TRK*.

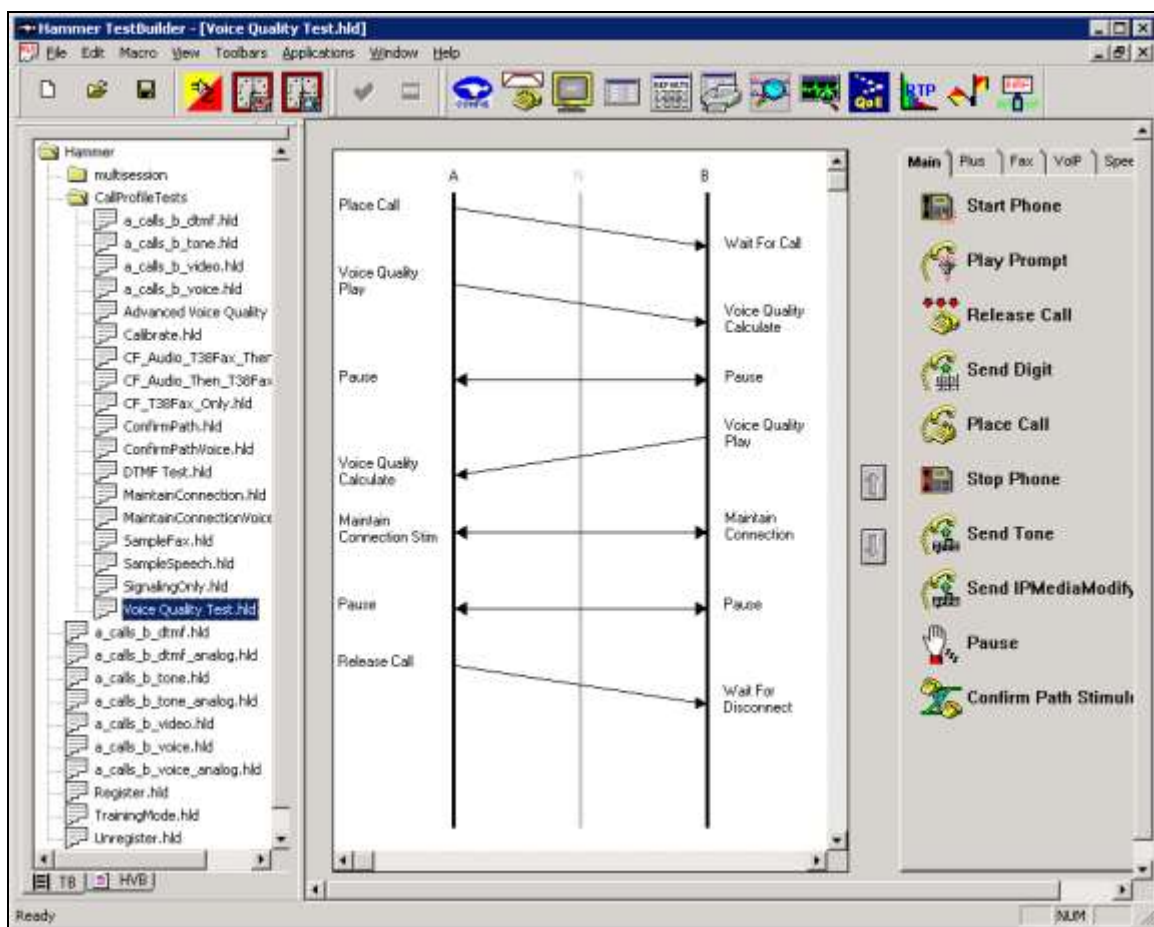


8.4 Configure and Run the Test Script

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

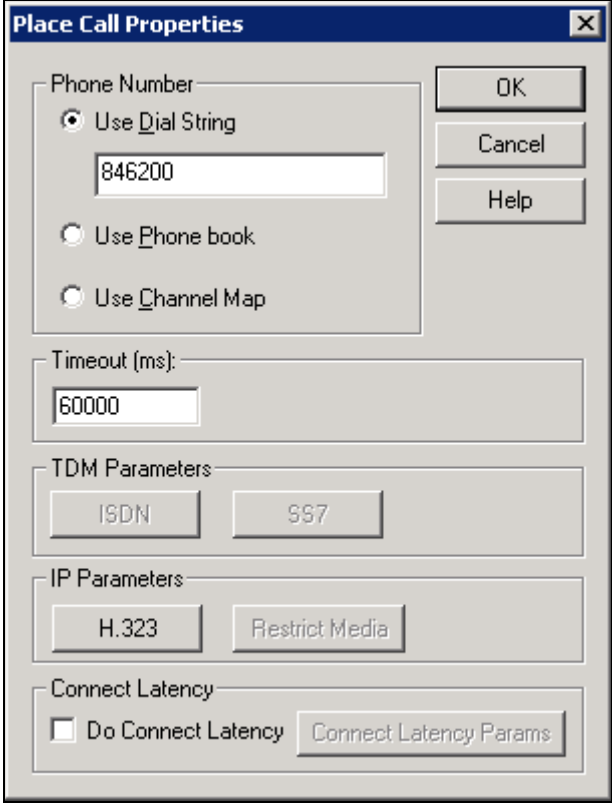
- a_calls_b_dtmf.hld to verify DTMF
- Voice Quality Test.hld to verify voice quality

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



In the sample test script configured above, the A-side (originating SIP trunk) places a call to the B-side (terminating SIP trunk) using the **Place Call** action. The **Place Call** properties can be configured by double-clicking on the action in the ladder diagram. In this example, Hammer IP dials the AAR access code '8' followed by "46200". The dial pattern should not match the **Trunk ANI** of the terminating Hammer SIP trunk.

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

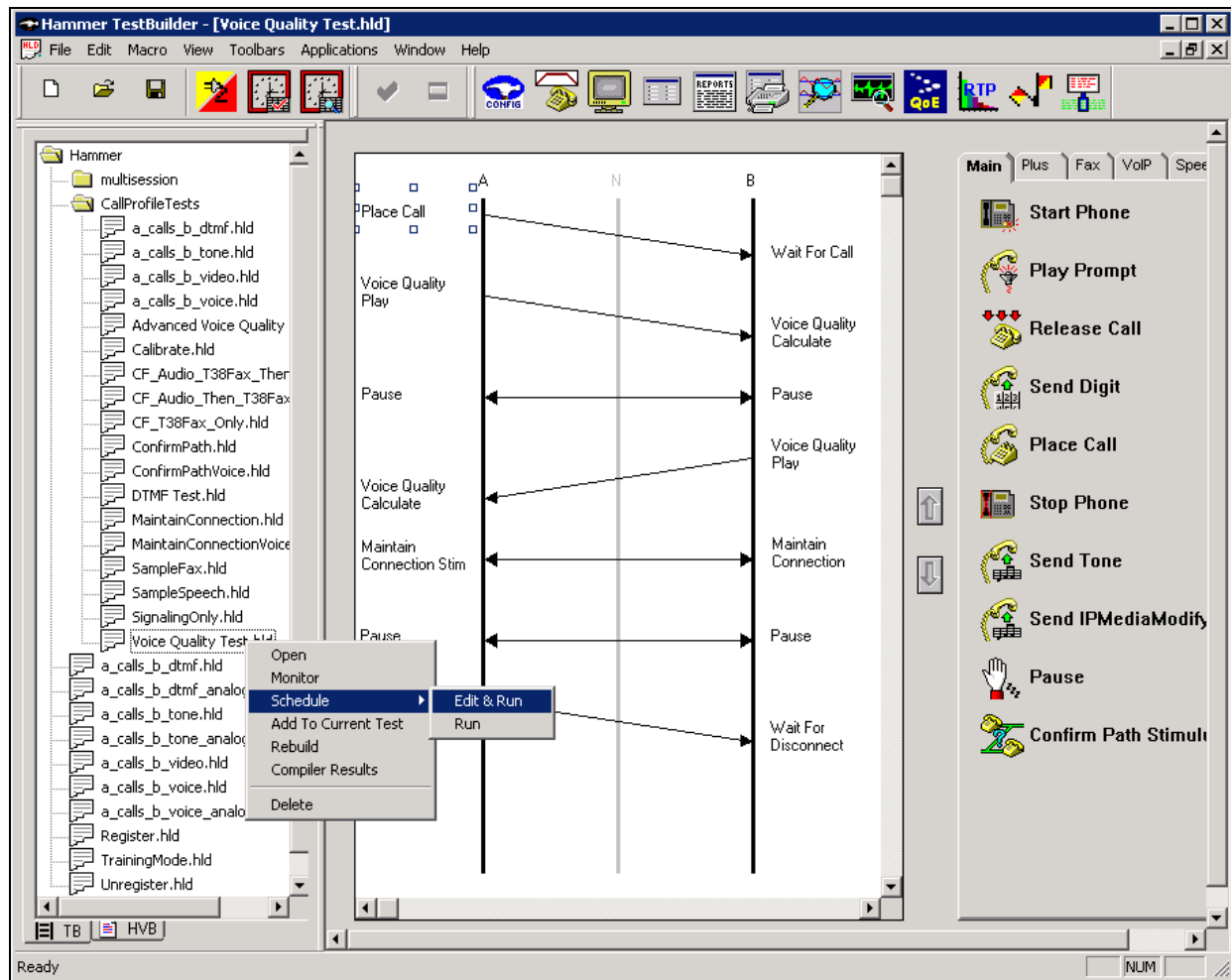


The image shows a 'Place Call Properties' dialog box with the following sections:

- Phone Number:** Contains three radio buttons: 'Use Dial String' (selected), 'Use Phone book', and 'Use Channel Map'. Below the radio buttons is a text field containing '846200'.
- Timeout (ms):** A text field containing '60000'.
- TDM Parameters:** Contains two buttons: 'ISDN' and 'SS7'.
- IP Parameters:** Contains two buttons: 'H.323' and 'Restrict Media'.
- Connect Latency:** Contains 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. For the compliance test, the **Stagger** time was set to *50 ms*.

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

Properties

TB Scheduler | Other

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

Start Time: 11:44:57 AM 11/24/2015

Action if a Channel is busy: Wait

Channels

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

B-Side: AVAYAEMPIRIX01[11-20]

PhoneBook

Select a PhoneBook: Default-phonebook

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

Stagger

☐ Automatic - Est. CHT (s) 5

☒ User Defined - (ms) 50

☐ Random - Min (s) 1 Max (s) 5

☐ None

OK Cancel Apply Help

9 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.

9.1 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 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 4
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: T00125	

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*, if it isn't, the field would be set to *ip-tdm* as shown below. Also, note that TLS port 5061 is being used.

status trunk 60/1	Page 2 of 4
CALL CONTROL SIGNALING	
Near-end Signaling Loc: PROCR	
Signaling IP Address	Port
Near-end: 192.168.100.10	: 5061
Far-end: 192.168.100.235	: 5061
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: MG1	Codec Type: G.711MU
Audio IP Address	Port
Near-end: 192.168.100.15	: 2066
Far-end: 192.168.100.170	: 10018
Video Near:	
Video Far:	
Video Port:	
Video Near-end Codec:	Video Far-end Codec:

Page 4 of the **status trunk** command indicates that SRTP is being used for the call.

```
status trunk 60/1                                     Page 4 of 4
SRC PORT TO DEST PORT TALKPATH
src port: T00044
T00044:TX:192.168.100.170:10018/g711u/20ms/1-srtp-aescm128-hmac80
001V057:RX:192.168.100.15:2066/g711u/20ms/1-srtp-aescm128-hmac80:TX:ctxID:394
001V060:RX:ctxID:394:TX:192.168.100.15:2060/g711u/20ms/1-srtp-aescm128-hmac80
T00125:RX:192.168.100.171:10006/g711u/20ms/1-srtp-aescm128-hmac80
dst port: T00125
```

9.2 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.

Session Manager Entity Link Connection Status

This page displays detailed connection status for all entity links from a Session Manager.

All Entity Links for Session Manager: tz-asn1

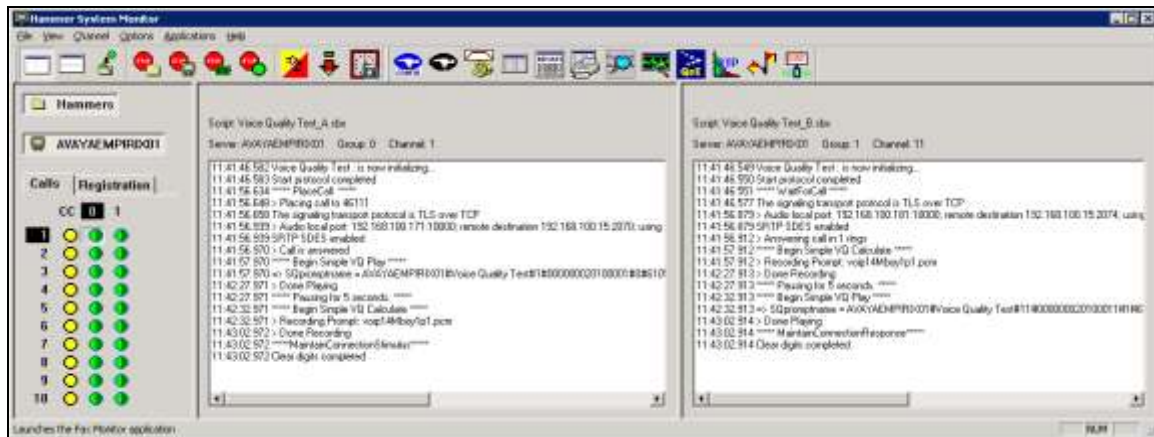
Status Details for the selected Session Manager:

Summary View

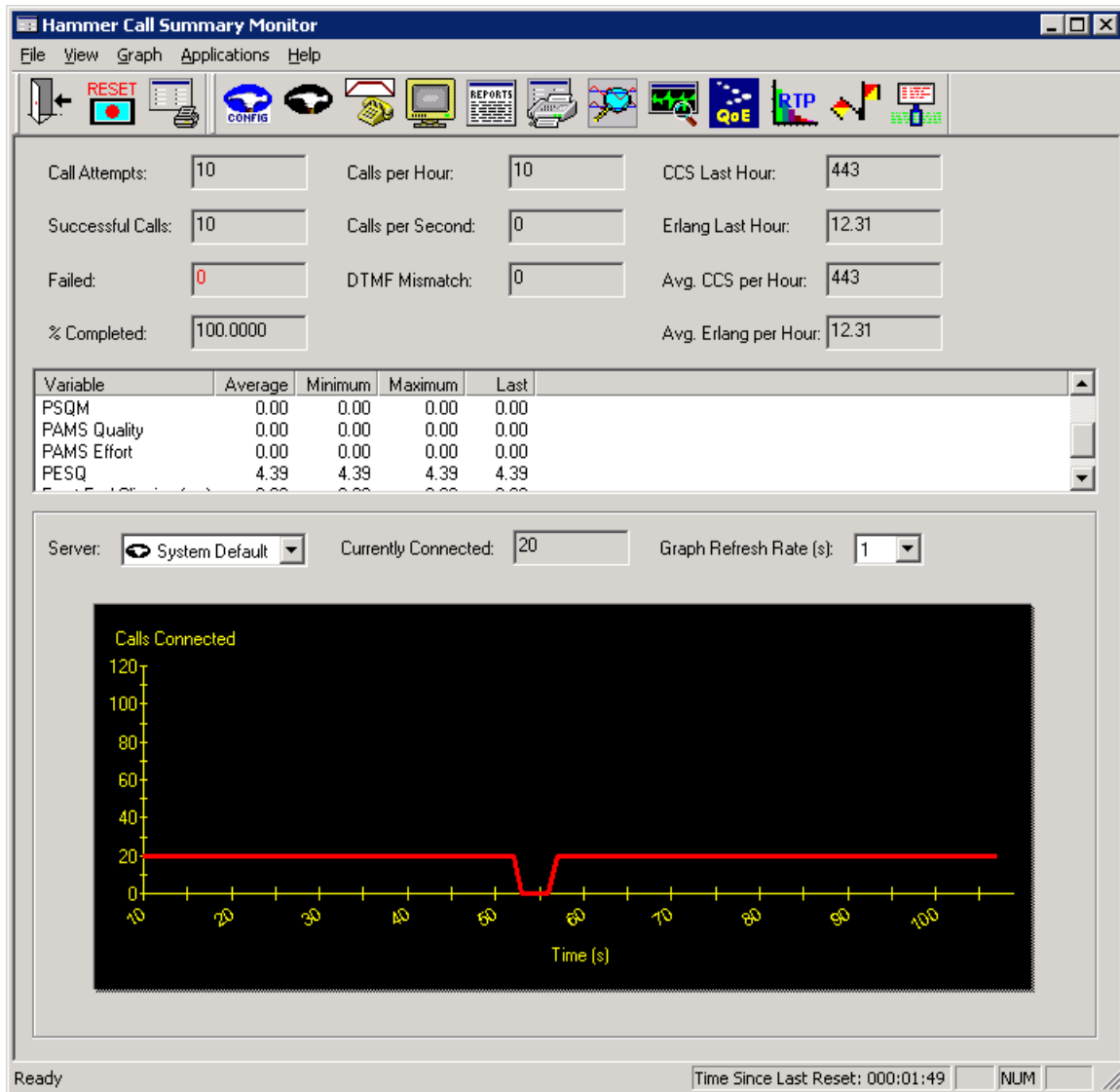
3 Items Refresh Filter: Enable

SIP Entity Name	SIP Entity Resolved IP	Port	Proto	Desc	Conn. Status	Reason Code	Link Status
Hammer-Inc	192.168.100.170	5061	TLS	FALSE	UP	200 OK	UP
devcon14	192.168.100.10	5061	TLS	FALSE	UP	200 OK	UP
Hammer-Out	192.168.100.171	5061	TLS	FALSE	UP	200 OK	UP

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. In the following System Monitor screen, it indicates that TLS over TCP and SRTP were being used for the test calls.



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.



10 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. Hammer IP was able to establish a SIP trunk with Avaya Aura® Session Manager, successfully establish calls through Avaya Aura® Communication Manager to SIP endpoints/trunks, generate voice quality metrics, monitor the calls, and generate reports. Furthermore, the solution was able to use SIPS to secure the SIP signaling using TLS (Transport Layer Security) and Secure Real-time Transport Protocol (SRTP) to protect the RTP data. All feature and serviceability test cases were completed successfully with observations noted in **Section 2.2**.

11 References

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

- [1] *Administering Avaya Aura® Communication Manager*, Release 6.3, Issue 10, August 2015, Document Number 03-300509.
- [2] *Administering Avaya Aura® System Manager for Release 6.3.13 through 6.3.15*, Release 6.3, Issue 8, December 2015.
- [3] *Administering Avaya Aura® Session Manager*, Release 6.3, Issue 7, September 2014.
- [4] *Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Endpoint Emulation with Secure SIP (SIPS) / Transport Layer Security (TLS)*, Issue 1.0, available at <http://www.avaya.com>.
- [5] *Empirix Hammer IP Installation Guide*, Release 6.2, October 2015, Revision A, 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 'rtptime'?	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.726 8 kb/s	Yes	122

Optional Descriptions

Include Field?	Field	Value
<input type="checkbox"/>	(i=) Media Information:	
<input type="checkbox"/>	(b=) Bandwidth Information:	

(a=) Attributes

Add Edit Delete

OK Cancel Help

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