



Avaya Solution & Interoperability Test Lab

Application Notes for Empirix Hammer G5 with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services using SIP Endpoint Emulation – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Empirix Hammer G5 test system with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services using SIP endpoint emulation. The Hammer G5 is a VoIP test platform consisting of a collection of applications used to configure the system, create, schedule, monitor tests, and create reports. In this configuration, the Hammer G5 emulates SIP endpoints that register with SIP Enablement Services. The Hammer G5 originates and terminates calls through the Avaya SIP telephony network. While the call is active, it 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 G5 can be used to verify the functionality, reliability and quality of VoIP services and applications running through the Avaya SIP telephony network.

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 G5 test system with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services using SIP endpoint emulation. The Hammer G5 is a VoIP test platform consisting of a collection of applications used to configure the system, create, schedule, monitor tests, and create reports. In this configuration, the Hammer G5 emulates SIP endpoints that register with SIP Enablement Services. The Hammer G5 originates and terminates calls through the Avaya SIP telephony network. While the call is active, it 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 G5 can be used to verify the functionality, reliability and quality of VoIP services and applications running through the Avaya SIP telephony network.

The following set of Hammer G5 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.

A solution using SIP trunk emulation on the Empirix Hammer G5 is described in [4] *Application Notes for Empirix Hammer G5 with Avaya Aura™ Communication Manager and Avaya Aura™ Communication Manager using SIP Trunk Emulation*.

1.1 Interoperability Compliance Testing

The interoperability compliance testing focused on verifying that the Hammer G5 can register with SIP Enablement Services as SIP endpoints, establish calls, send voice media, provide voice quality metrics, and generate reports. The following features and functionality were covered:

- SIP endpoint registration with SIP Enablement Services.
- Originating and terminating calls through the Avaya SIP telephony network.
- Support of G.711, G.723, 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 disabled was also verified.
- Generating voice quality metrics with Shuffling disabled.

1.2 Support

Technical support on the Empirix Hammer G5 can be obtained via phone, website, or email.

- **Phone:** (781) 266-3202
- **Web:** <http://www.empirix.com/support/maintenance.asp>
- **Email:** support@support.com

2 Reference Configuration

The network diagram below illustrates the test configuration. In this configuration, SIP Enablement Services receives calls from the Hammer G5, which emulates SIP endpoints. The call is routed through the Avaya SIP telephony network. The call is eventually routed back to the Hammer G5 where it is terminated. While the call is established, the Hammer G5 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 G5 applications running on the Hammer G5 server were used to configure the system, create and monitor the tests, and view the test reports.

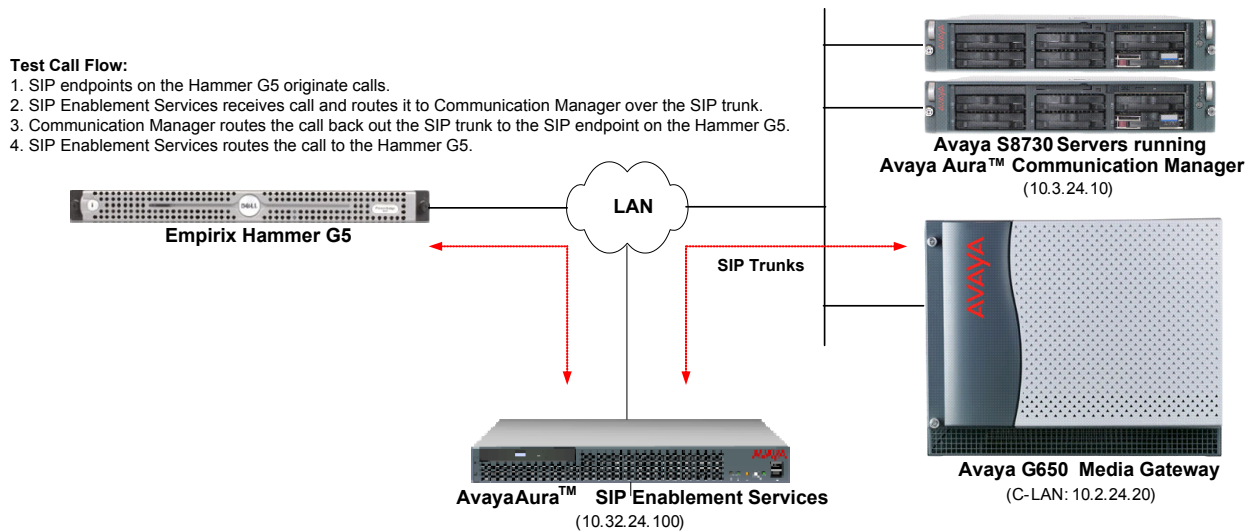


Figure 1: Empirix Hammer G5 with Avaya SIP Telephony Network

3 Equipment and Software Validated

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

Equipment	Software
Avaya S8730 Servers	Avaya Aura™ Communication Manager 5.2.1 (R015x.02.1.016.4) with Service Pack 1 (Patch 17959)
Avaya G650 Media Gateway <ul style="list-style-type: none">▪ TN799DP C-LAN Board▪ TN2302AP Media Processor Board	HW01 FW031 HW03 FW094
Avaya Aura™ SIP Enablement Services	Avaya Aura™ SIP Enablement Services 5.2.1 (SES-5.2.1.0-016.4)
Empirix Hammer G5 running on Microsoft Windows 2003 Server	1.5.8.295

4 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 SIP Enablement Services
- Administer SIP Stations

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

4.1 Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for the C-LAN board in the Avaya G650 Media Gateway and SIP Enablement Services. The host names will be used throughout the other configuration screens of Communication Manager.

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
Gateway001	10.32.24.1	
clancrm	10.32.24.20	
default	0.0.0.0	
medprocrm	10.32.24.21	
procr	0.0.0.0	
ses	10.32.24.100	
(6 of 6 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		

4.2 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 G5. 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. The Hammer G5 supports G.711, G.729AB, and G.723 codecs. For the compliance test, the IP codec set form was configured as shown below.

change ip-codec-set 1				Page 1 of 2
IP Codec Set				
Codec Set: 1				
Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)	
1: G.711MU	n	2	20	
2: G.729AB	n	2	20	
3: G.723-6.3K	n	1	30	

4.3 Administer IP Network Region

In the **IP Network Region** form, configure the codec set to be used for the 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 G650 Media Gateway. If voice quality metrics are to be generated by the test, disable Shuffling; otherwise, the audio traffic will not leave the Hammer G5 and the voice quality metrics will not yield accurate results. Shuffling can also be disabled at the station level. The **Authoritative Domain** for this configuration is *avaya.com*.

change ip-network-region 1		Page 1 of 19
IP NETWORK REGION		
Region: 1		
Location: Authoritative Domain: avaya.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? y
UDP Port Max: 3029		
DIFFSERV/TOS PARAMETERS		RTCP Reporting Enabled? y
Call Control PHB Value: 34		RTCP MONITOR SERVER PARAMETERS
Audio PHB Value: 46		Use Default Server Parameters? y
Video PHB Value: 26		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 7		
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		

4.4 Administer SIP Trunk Group to SIP Enablement Services

Prior to configuring a SIP trunk group for to SIP Enablement Services, a SIP signaling group must be configured. Configure the **Signaling Group** form as shown below. The **Near-end Node Name** and **Far-end Node Name** are the C-LAN board and SIP Enablement Services, respectively.

add signaling-group 10		Page 1 of 1
SIGNALING GROUP		
Group Number: 10	Group Type: sip	
	Transport Method: tls	
IMS Enabled? n		
Near-end Node Name: clancrm	Far-end Node Name: ses	
Near-end Listen Port: 5061	Far-end Listen Port: 5061	
	Far-end Network Region: 1	
Far-end Domain: avaya.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? y	
Enable Layer 3 Test? n	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Direct IP-IP Early Media? n	
	Alternate Route Timer(sec): 6	

Configure the **Trunk Group** form as shown below. 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. All calls from the Hammer G5 will be routed over this trunk group. Each call will use two trunk members one for the incoming call and another one for the outgoing call.

add trunk-group 10		Page 1 of 21
TRUNK GROUP		
Group Number: 10	Group Type: sip	CDR Reports: y
Group Name: To SES	COR: 1	TN: 1 TAC: 1010
Direction: two-way	Outgoing Display? n	
Dial Access? n	Night Service:	
Queue Length: 0		
Service Type: tie	Auth Code? n	
Signaling Group: 10		
Number of Members: 60		

4.5 Administer SIP Stations

Configure a SIP station for each SIP channel on the Hammer G5. Use the default value for the **Type**, which is *6408D+*. Set the **Port** field to *X* and configure a descriptive **Name**. For the compliance test, 20 SIP stations were used with extensions ranging from 25100 to 25119. The first group of 10 channels (extensions 25100 to 25109) was used to originate calls. The calls were then terminated on the remaining 10 channels (extensions 25110 to 25119). Repeat this procedure for each channel required by the Hammer test.

add station 25100		Page 1 of 5
STATION		
Extension: 25100	Lock Messages? n	BCC: 0
Type: 6408D+	Security Code:	TN: 1
Port: X	Coverage Path 1:	COR: 1
Name: Hammer SIP 0	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
	Time of Day Lock Table:	
Loss Group: 2	Personalized Ringing Pattern: 1	
Data Module? n	Message Lamp Ext: 25100	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english		
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? n	

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

add station 25100		Page 2 of 5
STATION		
FEATURE OPTIONS		
LWC Reception: spe	Auto Select Any Idle Appearance? n	
LWC Activation? y	Coverage Msg Retrieval? y	
LWC Log External Calls? n	Auto Answer: none	
CDR Privacy? n	Data Restriction? n	
Redirect Notification? y	Idle Appearance Preference? n	
Per Button Ring Control? n	Bridged Idle Line Preference? n	
Bridged Call Alerting? n	Restrict Last Appearance? y	
Active Station Ringing: single		
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
Service Link Mode: as-needed	EC500 State: enabled	
Multimedia Mode: basic	Audible Message Waiting? n	
MWI Served User Type:	Display Client Redirection? n	
AUDIX Name:	Select Last Used Appearance? n	
	Coverage After Forwarding? s	
	Multimedia Early Answer? n	
	Direct IP-IP Audio Connections? y	
Emergency Location Ext: 25100	IP Audio Hairpinning? n	

Configure the **Stations with Off-PBX Telephone Integration** form so that calls destined for a SIP endpoint on the Hammer G5 are routed to SIP Enablement Services, which will then route the call to the Hammer G5. On this form, specify the extension of the SIP endpoint and set the **Application** field to *OPS*. The **Phone Number** field is set to the digits to be sent over the SIP trunk. In this case, the SIP telephone extensions configured on SIP Enablement Services also match the extensions of the corresponding stations on Communication Manager. However, this is not a requirement. Finally, the **Trunk Selection** field is set to *10*, the SIP trunk group number. This field specifies the trunk group used to route the outgoing call. Another option for routing a call over a SIP trunk group is to use Auto Alternate Routing (AAR) or Auto Route Selection (ARS) routing instead. In this case, the **Trunk Selection** field would be set to *aar* or *ars*. Configuration of other AAR or ARS forms would also be required. Refer to Error! Reference source not found. for information on routing calls using AAR or ARS. Repeat this step for each SIP endpoint required on the Hammer G5 (e.g., extensions 25100 to 25119).

change off-pbx-telephone station-mapping 25100							Page 1 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
25100	OPS	-		25100	10	1	
		-					
		-					
		-					

5 Configure Avaya Aura™ SIP Enablement Services

This section covers the administration of SIP Enablement Services. SIP Enablement Services is configured via an Internet browser using the Administration web interface. To access the Administration web interface, enter *http://<ip-addr>/admin* as the URL in an Internet browser, where *<ip-addr>* is the IP address of SIP Enablement Services. Log in with the appropriate credentials and then select **Administration→SIP Enablement Services** from the next screen. The main screen is displayed as shown below.

AVAYA

Integrated Management
SIP Server Management

Help Exit

This Server: [1] ses1

Top

Users

Address Map Priorities

Adjunct Systems

Aggregator

Certificate Management

Conferences

Emergency Contacts

Export/Import to ProVision

Hosts

IM logs

Communication Manager Servers

Communication Manager Extensions

Server Configuration

SIP Phone Settings

Survivable Call Processors

System Status

Trace Logger

Trusted Hosts

Top

Manage Users	Add and delete Users.
Manage Address Map Priorities	Adjust Address Map Priorities.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Event Aggregators	Add/Delete Event Aggregators.
Certificate Management	Manage Certificates.
Manage Conferencing	Add and delete Conference Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Export Import to ProVision	Export and import data using ProVision on this host.
Manage Hosts	Add and delete Hosts.
IM logs	Download IM Logs.
Manage Communication Manager Servers	Add and delete Communication Manager Servers.
Manage Communication Manager Extensions	Add and delete Communication Manager Extensions.
Server Configuration	View Properties of the system.
Manage SIP Phone Settings	Add/Delete Phone Settings
Manage Survivable Call Processors	Add and delete Survivable Call Processors.
System Status	View System Status.
Trace Logger	Manage SIP Trace Logs.
Manage Trusted Hosts	Add and delete Trusted Hosts.

From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. In the **View System Properties** screen, enter the **SIP Domain** name assigned to the Avaya SIP-based network and the **SIP License Host**. For the **SIP License Host** field, enter the fully qualified domain name or the IP address of the SIP Enablement Services server that is running the WebLM application and has the associated license file installed. This entry should always correspond to the localhost unless the WebLM server is not co-resident with this server. After configuring the **View System Properties** screen, click the **Update** button.

The screenshot shows the Avaya Integrated Management SIP Server Management web interface. The top header includes the Avaya logo, the title "Integrated Management SIP Server Management", and a status bar indicating "This Server: [1] ses1". A navigation menu on the left lists various configuration options, with "System Properties" selected under "Server Configuration". The main content area is titled "View System Properties" and displays several configuration fields:

- System Information:**
 - SES Version: SES-5.2.1.0-016.4
 - System Configuration: Simplex
 - Host Type: SES combined home-edge
- SIP Domain:**
 - SIP Domain*:
 - Note that the DNS domain is avaya.com
 - Help text: If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com
- SIP License Host:**
 - SIP License Host*:
- DiffServ/TOS Parameters:**
 - Call Control PHB Value*:
- 802.1 Parameters:**
 - Priority Value*:
 - Management System Access Login:
 - Management System Access Password:
 - DB Log Level:

An "Update" button is located at the bottom left of the configuration area.

After setting up the domain in the **System Properties** screen, create a host entry for SIP Enablement Services. The following example shows the **Edit Host** screen since the host had already been configured. Enter the IP address of SIP Enablement Services in the **Host IP Address** field. The **Profile Service Password** was specified during the system installation. Next, configure the **Host Type** field. In this example, the host server was configured as an *SES combined home-edge*. The default values for the other fields may be used as shown below. Click the **Update** button.

AVAYA Integrated Management SIP Server Management
 Help Exit This Server: [1] ses1

Top

- Users
 - Add
 - Default Profile
 - Delete
 - Edit
 - List
 - Password
 - Search
 - Manage All Registered Users
 - Search Registered Devices
 - Search Registered Users
- Address Map Priorities
- Adjunct Systems
- Aggregator
- Certificate Management
- Conferences
- Emergency Contacts
- Export/Import to ProVision
- Hosts
 - IM logs
 - Communication Manager Servers
 - Communication Manager Extensions
 - Server Configuration
 - SIP Phone Settings
 - Survivable Call Processors
 - System Status
 - Trace Logger
 - Trusted Hosts

Edit Host

Host IP Address* 10.32.24.100

Profile Service Password*

Host Type SES combined home-edge

Parent none

Listen Protocols ☒ UDP ☒ TCP ☒ TLS

Link Protocols ☐ UDP ☐ TCP ☒ TLS

Access Control Policy (Default) ☐ Allow All ☒ Deny All

Emergency Contacts Policy ☒ Allow ☐ Deny

Minimum Registration (seconds) 900 Registration Expiration Timer (seconds)* 900

Subscription Expiration Timer (seconds)* 86400

Line Reservation Timer (seconds) 30

Outbound Routing Allowed ☒ Internal ☒ External

From

OutboundProxy Port ☐ UDP ☐ TCP ☐ TLS

Outbound Direct Domains

Default Ringer Volume* 5 Default Ringer Cadence 2

Default Receiver Volume* 5 Default Speaker Volume* 5

VMM Server Address

VMM Server Port 5005 VMM Report Period 5

Fields marked * are required.

Update

Under the **Communication Manager Servers** option in the Administration web interface, select **Add** to add the Avaya S8730 Servers since a SIP trunk is required between Communication Manager and SIP Enablement Services. In the **Add Communication Manager Server Interface** screen, enter the following information:

- A descriptive name in the **Communication Manager Server Interface Name** field (e.g., *devcon13*).
- Select the home server in the **Host** field.
- Select *TLS* (Transport Link Security) for the **SIP Trunk Link Type**. TLS provides encryption at the transport layer.
- Enter the IP address of the C-LAN board in the Avaya G650 Media gateway in the **SIP Trunk IP Address** field.

Refer to [3] for additional information on configuring the remaining fields. Click the **Update** button.

The screenshot displays the Avaya Integrated Management SIP Server Management web interface. The top navigation bar includes the Avaya logo, 'Integrated Management SIP Server Management', and 'This Server: [1] ses1'. A left sidebar contains a menu with options like 'Users', 'Adjunct Systems', 'Aggregator', 'Certificate Management', 'Conferences', 'Emergency Contacts', 'Export/Import to ProVision', 'Hosts', 'IM logs', 'Communication Manager Servers', 'Communication Manager Extensions', 'Server Configuration', 'SIP Phone Settings', and 'Survivable Call Processors'. The main content area is titled 'Edit Communication Manager Server Interface' and contains the following fields:

- Communication Manager Server Interface Name***: devcon13
- Host**: 10.32.24.100
- SIP Trunk**
 - SIP Trunk Link Type**: ☐ TCP ☒ TLS
 - SIP Trunk IP Address***: 10.32.24.20
- Communication Manager Server**
 - Communication Manager Server Admin Address* (see Help)**: 10.32.24.10
 - Communication Manager Server Admin Port***: 5022
 - Communication Manager Server Admin Login***: interop
 - Communication Manager Server Admin Password***: [masked]
 - Communication Manager Server Admin Password Confirm***: [masked]
- SMS Connection Type**: ☒ SSH ☐ Telnet ☐ Not Available

A note at the bottom states: 'Note: If the Communication Manager Server connection type is changed and the admin port value is not also changed, changing connection type to SSH will change the admin port to 5022 when Add or Update is clicked and changing connection type to Telnet will change admin port to 5023 when Add or Update is clicked.' Below the fields, it says 'Fields marked * are required.' and there is an 'Update' button.

Add an user for each SIP endpoint required on the Hammer G5. In the **Add User** screen, enter the extension of the SIP endpoint in the **Primary Handle** field. Enter a user password in the **Password** and **Confirm Password** fields. In the **Host** field, select the SIP Enablement Services server hosting the domain (avaya.com) for this user. Enter the **First Name** and **Last Name** of the user. To associate a Communication Manager server extension with this user, select the **Add Communication Manager Extension** checkbox. Calls from this user will always be routed through Communication Manager over the SIP trunk for origination services. The **Add Communication Manager Extension** screen will be displayed after adding this user profile by clicking on the **Add** button.

AVAYA Integrated Management SIP Server Management
 Help Exit This Server: [1] ses1

Top
 Users
 Add
 Default Profile
 Delete
 Edit
 List
 Password
 Search
 Manage All Registered Users
 Search Registered Devices
 Search Registered Users
 Address Map Priorities
 Adjunct Systems
 Aggregator
 Certificate Management
 Conferences
 Emergency Contacts
 Export/Import to ProVision
 Hosts
 IM logs
 Communication Manager Servers
 Communication Manager Extensions

Add User

Primary Handle* 25100
 User ID
 Password*
 Confirm Password*
 Host* 10.32.24.100
 First Name* Hammer
 Last Name* SIP0
 Address 1 211 Mount Airy Rd
 Address 2
 Office
 City Basking Ridge
 State NJ
 Country USA
 Zip 07920
 Survivable Call Processor none
 Add Communication Manager Extension ☒
 Fields marked * are required.

Add

In the **Add Communication Manager Extension** screen, enter the **Extension** configured on the Communication Manager for the previously added user. Usually, the Communication Manager extension and the user extension are the same (recommended). Select the **Communication Manager Server** assigned to this extension. Click the **Add** button.



The screenshot displays the Avaya Integrated Management SIP Server Management web interface. The header includes the Avaya logo, the title "Integrated Management SIP Server Management", and a status bar indicating "This Server: [1] ses1". A navigation menu on the left lists options such as "Top", "Users", "Add", "Default Profile", "Delete", "Edit", "List", "Password", "Search", "Manage All Registered Users", "Search Registered Devices", and "Help Exit". The main content area is titled "Add Communication Manager Extension" and contains the following text: "Add Communication Manager extension for user 25200.", "Extension" (with a text input field containing "25100"), "Communication Manager Server" (with a dropdown menu showing "devcon13"), and a note "Fields marked * are required." Below the form is an "Add" button.

6 Configure Empirix Hammer G5

This section provides the procedures for configuring the Empirix Hammer G5. The procedures fall into the following areas:

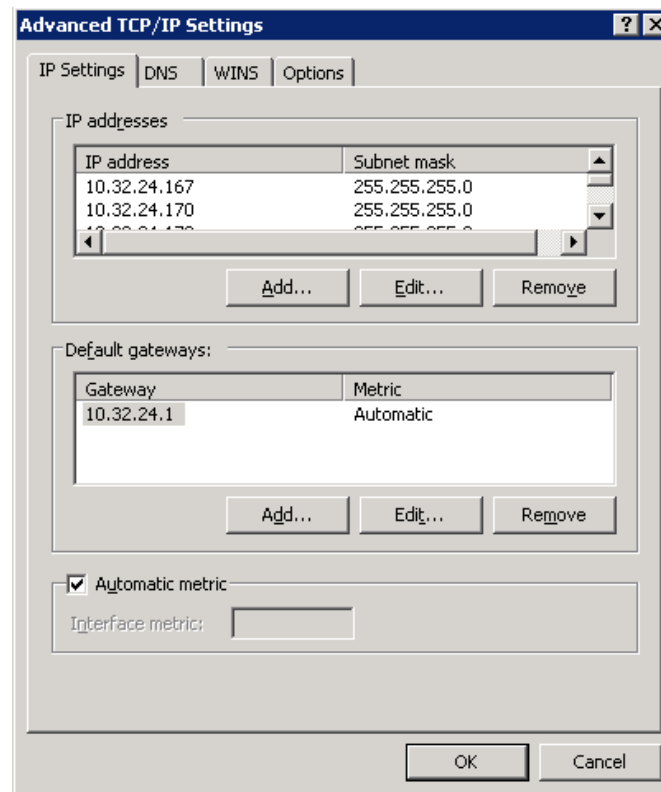
- Configure IP addresses to be used by each Hammer G5 channel.
- Configure the system, including the SIP endpoint interface and the phone book, using the **Hammer Configurator**.
- Create and run the test script using the **Hammer TestBuilder**.

Monitoring test call progress is covered in the **Verify Empirix Hammer G5** section (Section 8.3).

Note: Refer to [5] for instructions on installing the Hammer G5 software.

6.1 Configure IP Addresses on Hammer G5 Server

The Hammer G5 server needs to be configured with the IP addresses that it will need for each channel. During the compliance test, 20 SIP endpoint channels were used. 10 channels were used to originate calls and 10 channels were used to terminate calls. This requires a block of 20 IP addresses, which must be contiguous. The 20 IP addresses used ranged from 10.32.24.160 to 10.32.24.179. These IP addresses are configured in the **Advanced TCP/IP Settings** under Network Connections in Windows 2003 Server.

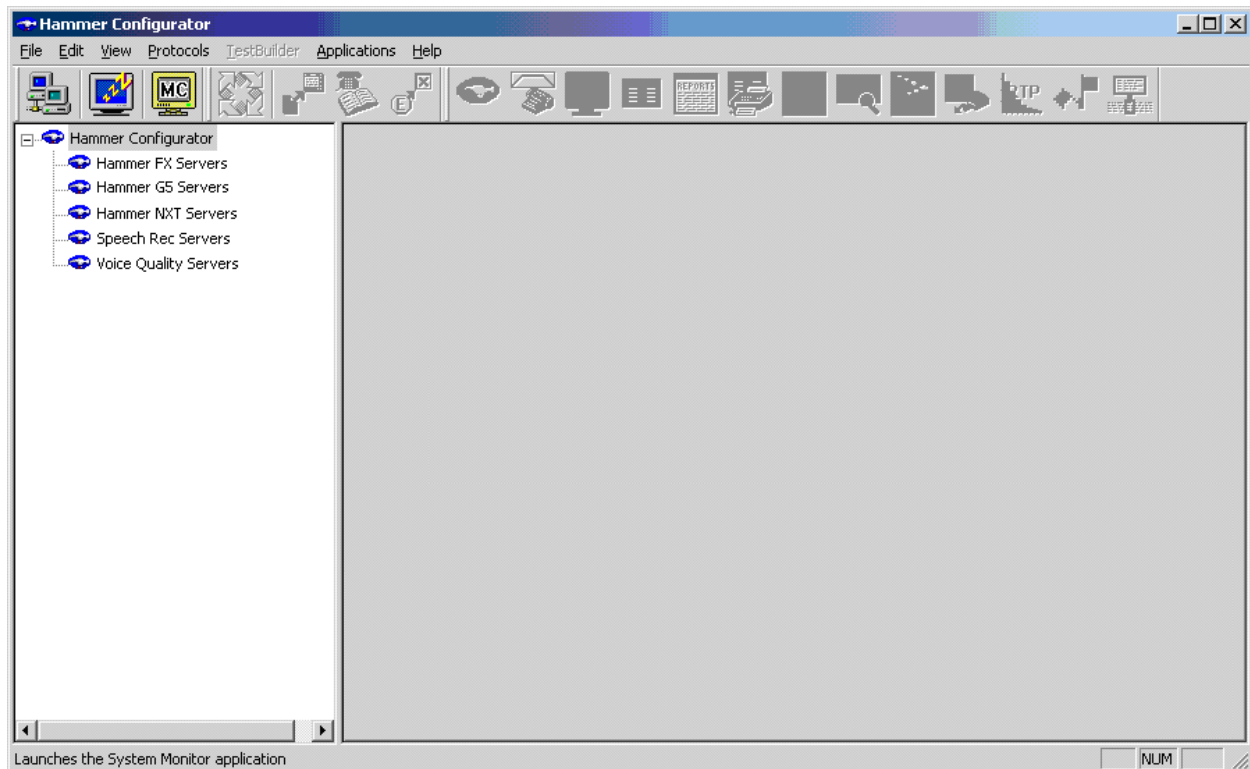


6.2 Configure the System

This section covers the configuration of the SIP endpoint interface and the phone book.

6.2.1 Configure SIP Endpoint Interface

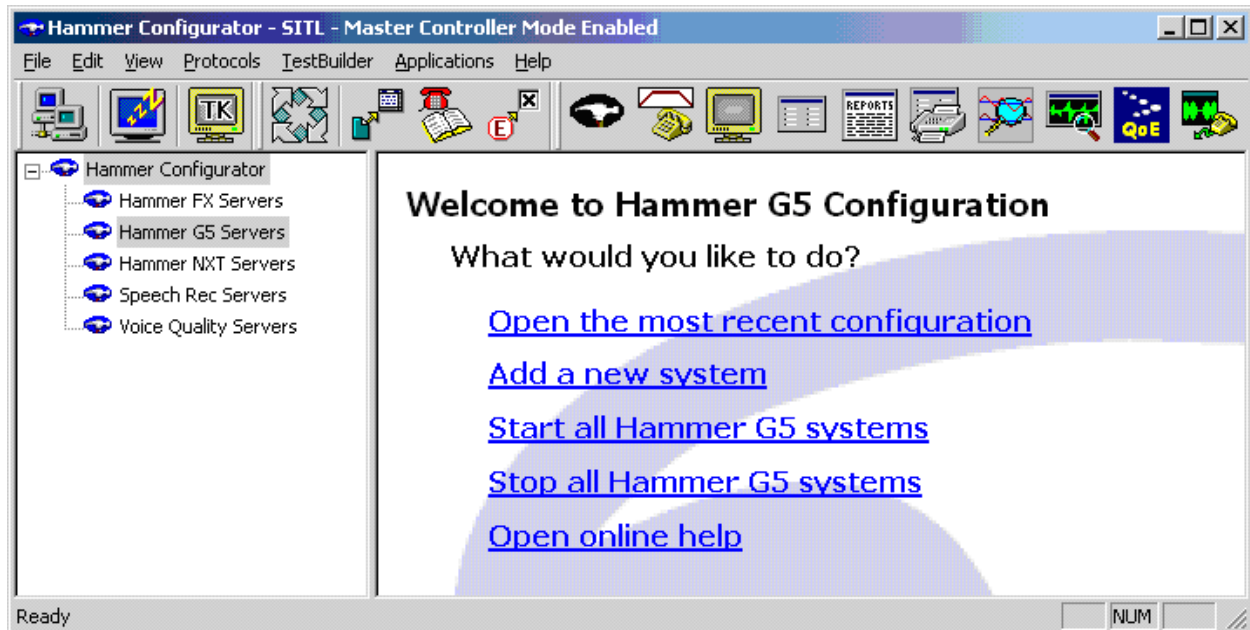
The Empirix Hammer G5 is configured through the **Hammer Configurator**, a graphical user interface, residing on the Hammer G5 server. From the Hammer G5 server, run the **Hammer Configurator**. The initial window is displayed as follows.



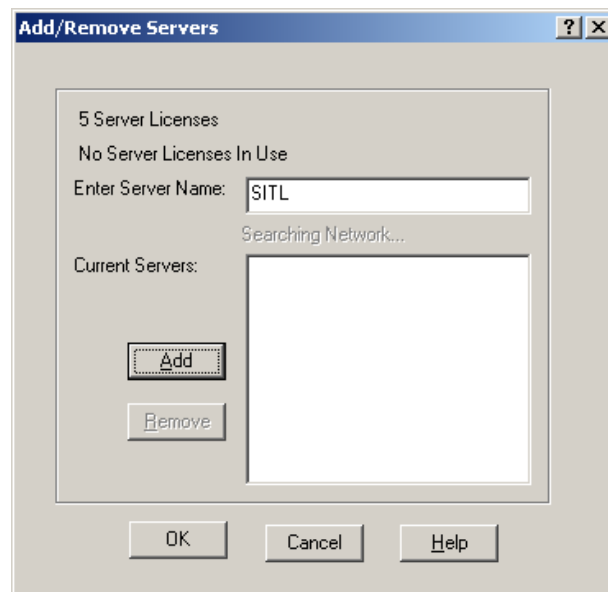
The Hammer G5 must be in **Master Controller Mode**. Verify that it is in Master Controller

Mode by clicking the  icon. If the  icon is not displayed, the Hammer G5 is already in Master Controller Mode.

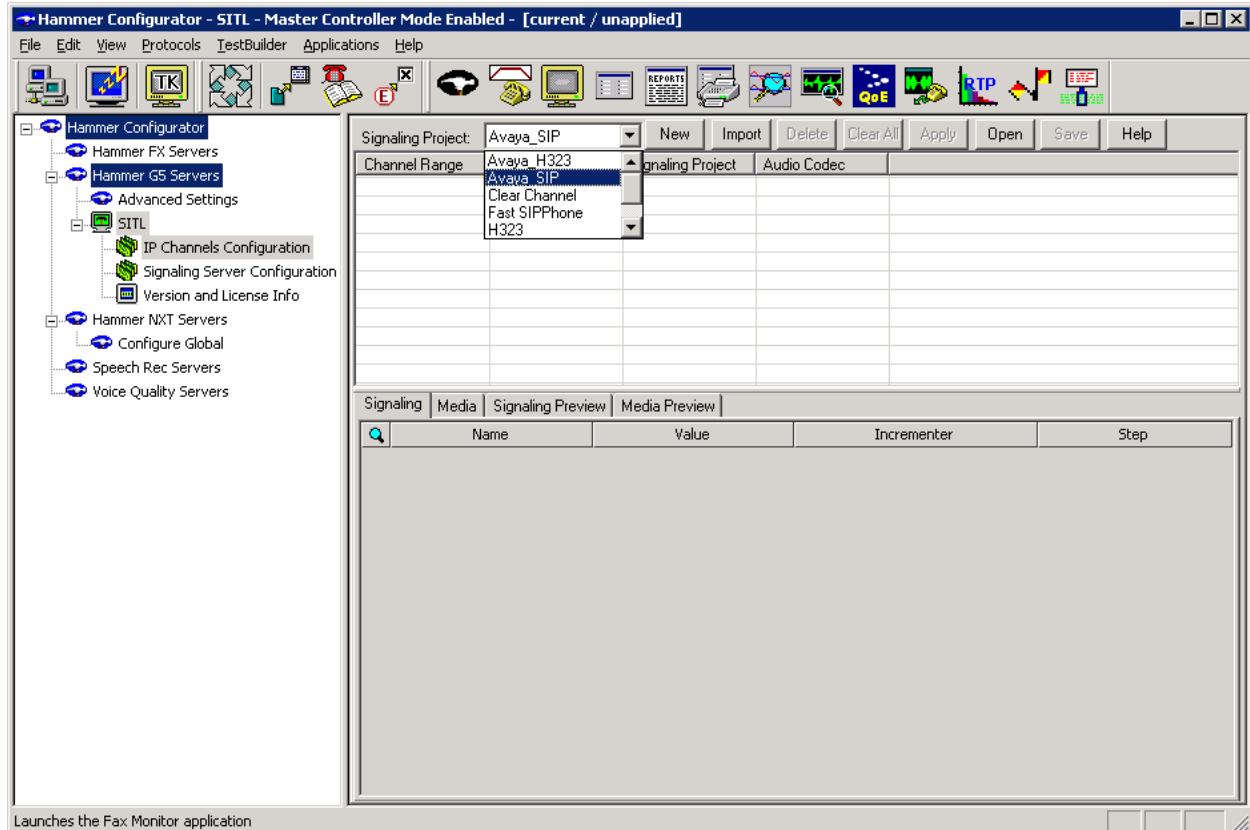
Next, select Hammer G5 Servers item in the left pane and then click on **Add a new system**. The **Add/Remove Servers** window will be displayed.



In the **Add/Remove Servers** window, specify a descriptive **Server Name** and then click the **Add** button. The server name will be moved to the **Current Servers** section. Click **OK**.

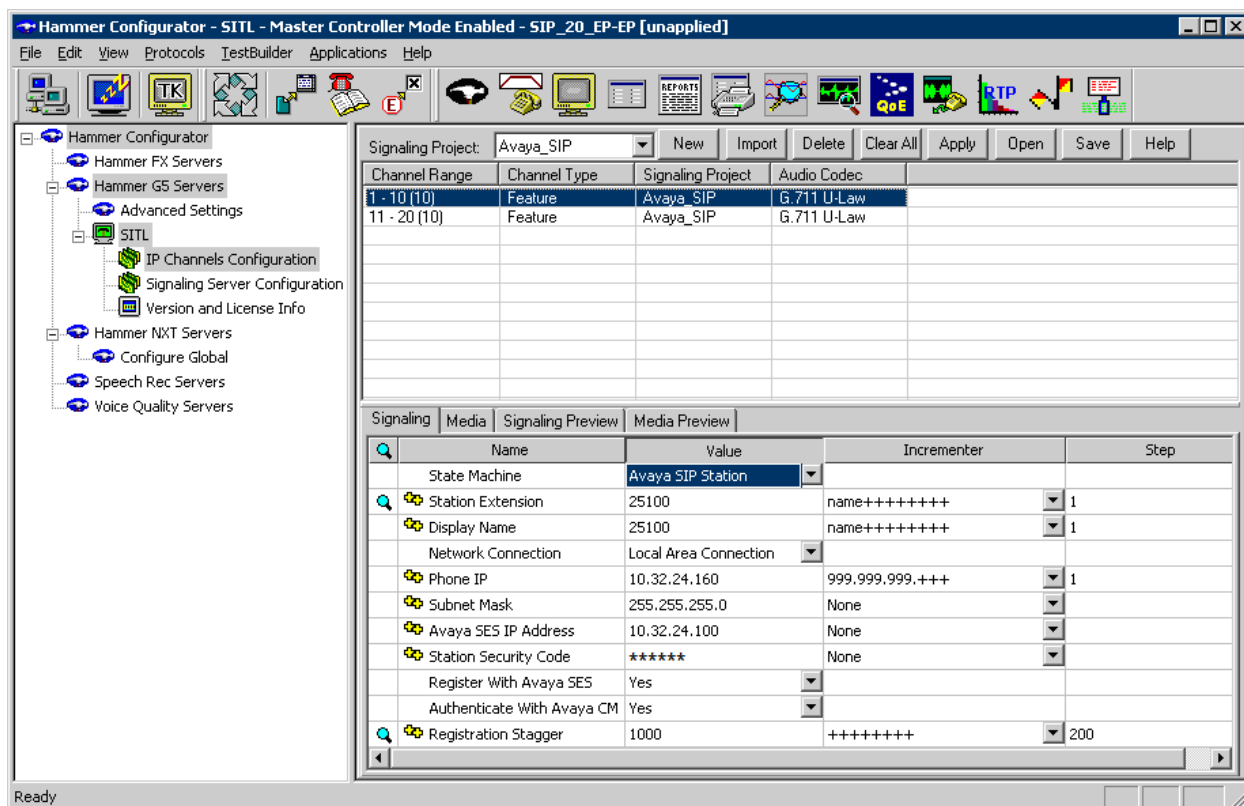


Once the server is added to the system, the server name will appear in the left pane of the **Hammer Configurator**. Expand the server name (e.g., *SITL*) in the left pane and click on *IP Channels Configuration*. The following window will be displayed. Select *Avaya SIP* for the **Signaling Project** 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 then be set as follows:

- **State Machine** should be set to *Avaya SIP Station*.
- **Station Extension** should be set to the first extension in the group (e.g., *25100*) and the **Incrementer** and **Step** fields should be set as shown so that the extension of the subsequent channels are incremented by one. This covers extensions from 25100 to 25109.
- **Network Connection** should be set to the appropriate network interface on the server.
- **Phone IP** should be set to the IP address of the first channel in the group and the **Incrementer** and **Step** fields should be set as shown so that the last of octet of the IP address is incremented by one. Note that this requires a block of contiguous IP addresses. This covers IP addresses from *10.32.24.160* to *10.32.24.169*.
- **Subnet Mask** should be set to the network mask (e.g., *255.255.255.0*).
- **Avaya SES IP Address** should be set to the IP address of SIP Enablement Services.
- **Station Security Code** should match the one configured in the corresponding **User** form on SIP Enablement Services.
- **Register with Avaya SES** should be set to *Yes*.
- **Authenticate with Avaya CM** should be set to *Yes*.



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, G729, and G.723 codecs are supported.
- **Frequency [ms]** should be set to the appropriate value for the specified codec. It should match the Packet Size [ms] field in the **IP Codec Set** form on Communication Manager for the specified codec.
- **Network Connection** should specify the appropriate network interface.
- **Source IP Address** should be set to the IP address of the first channel in the group. The **Incrementer** and **Step** fields should be set as shown so that the last octet of the IP address is incremented for the subsequent channels. Note that the IP addresses for the channels need to be contiguous.
- **Media Profile** should be set to one that specifies the codec configured in the **Audio Codec** field.
- The remaining fields should be configured as shown.

Hammer Configurator - SITL - Master Controller Mode Enabled - SIP_20_EP-EP [unapplied]

File Edit View Protocols TestBuilder Applications Help

Hammer Configurator

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

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

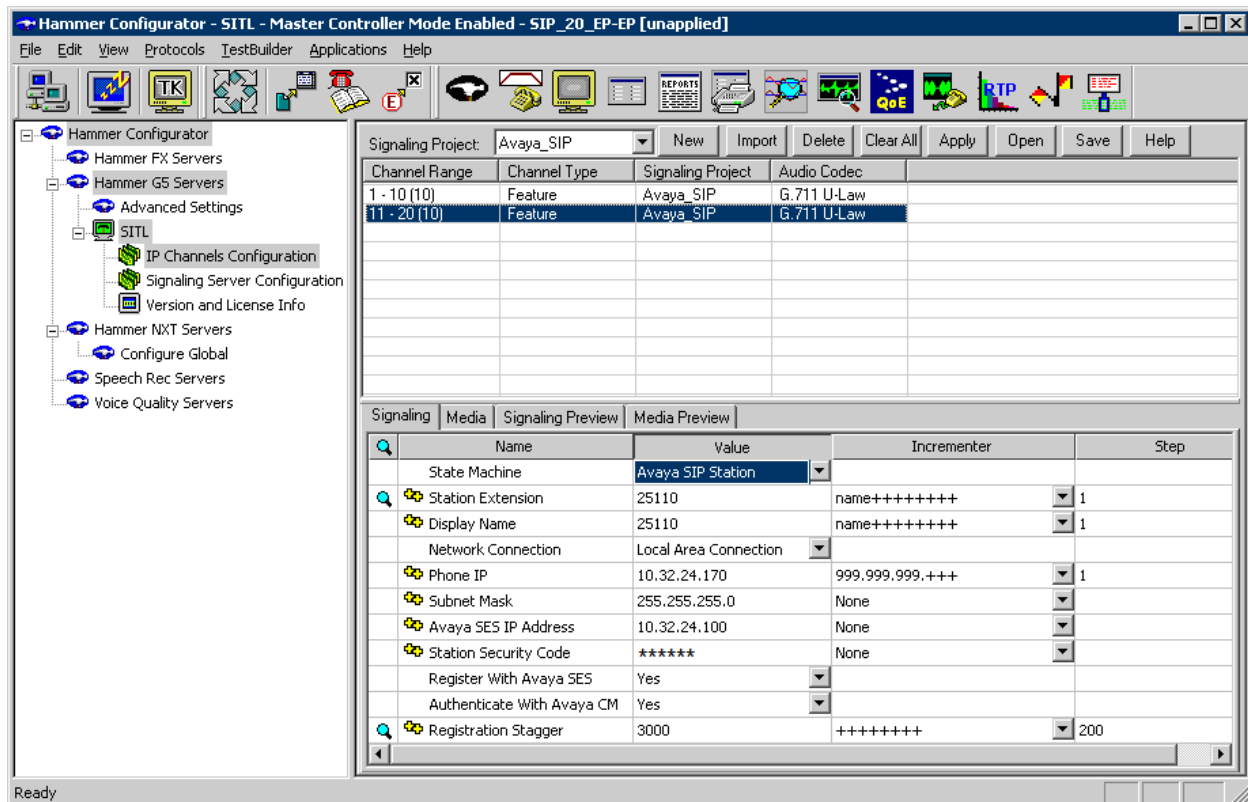
Channel Range	Channel Type	Signaling Project	Audio Codec
1 - 10 (10)	Feature	Avaya_SIP	G.711 U-Law
11 - 20 (10)	Feature	Avaya_SIP	G.711 U-Law

Signaling Media Signaling Preview Media Preview

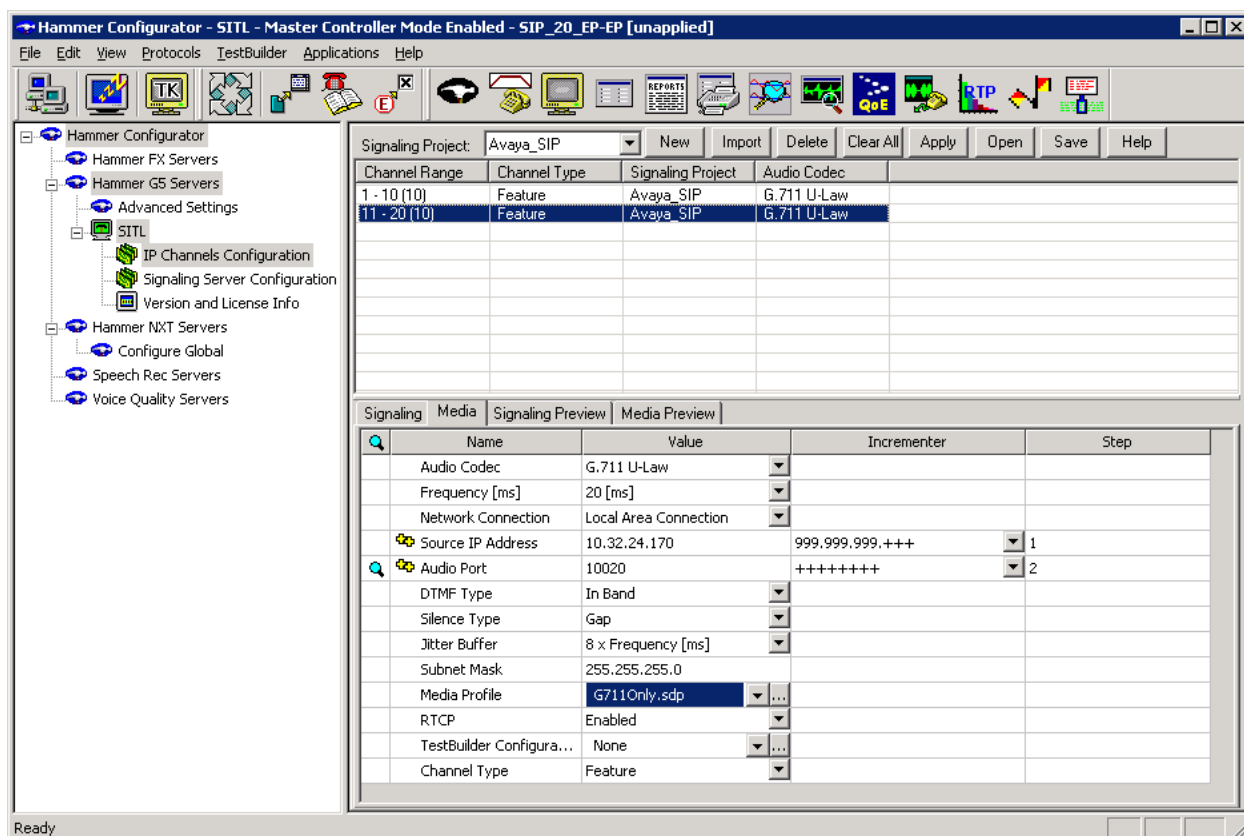
Name	Value	Incrementer	Step
Audio Codec	G.711 U-Law		
Frequency [ms]	20 [ms]		
Network Connection	Local Area Connection		
Source IP Address	10.32.24.160	999.999.999.+++	1
Audio Port	10000	+++++++	2
DTMF Type	In Band		
Silence Type	Gap		
Jitter Buffer	8 x Frequency [ms]		
Subnet Mask	255.255.255.0		
Media Profile	G711Only.sdp		
RTCP	Enabled		
TestBuilder Configuration	None		
Channel Type	Feature		

Ready

The second line in the grid that is highlighted in the figure below corresponds to the second group of channels that will terminate the 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 with the exception that the **Station Extension** and **Phone IP** fields will be different. This group of channels will be assigned extensions 25110 to 25119 and IP addresses from 10.32.24.170 to 10.32.24.179. Again, the IP addresses for this group of channels must be contiguous.



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.



The following windows show the configuration of the **Media Profile** used in the **Media** tab above for the two channel groups. To access this window, click on the ellipses button (...) in the **Media** tab. By default, there are **Media Profiles** for each audio codec type so this configuration is not required since the profiles are available out-of-box. It is presented for informational purposes. Click on the **Audio Description** button to view the codecs that will be advertised by the Hammer G5 when placing a call.

The screenshot shows the 'Media Profile Editor' window with the title bar path: \\SITL\Hammer\IPSigServer\SDPs\G711Only.sdp. The window is divided into three main sections:

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

Include Field?	Field	Value
<input type="checkbox"/>	(o=) Owner:	HammerG5
<input type="checkbox"/>	(s=) Session Name:	Hammer 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 for adding attributes, with 'Add', 'Edit', and 'Delete' buttons to its right.
- Media Descriptions:** Three checkboxes: 'Audio Description' (checked), 'Image (T.38) Description' (unchecked), and 'Video Description' (unchecked).

At the bottom of the window 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 G711Only.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.

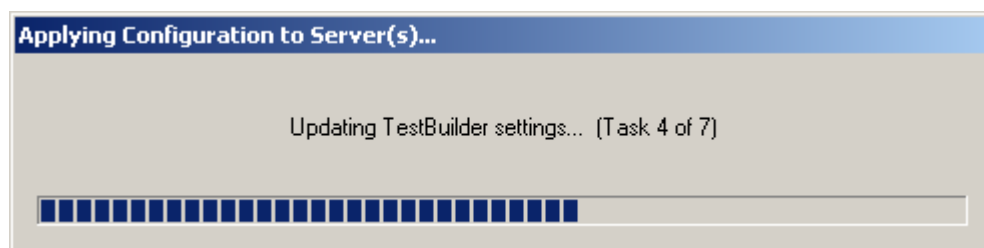
The screenshot shows a dialog box titled "MPE Audio Description: \\SITL\Hammer\IP5igServer\SDPs\G711Only.sdp". The main section is titled "Order and configure codecs to advertise in Media Profile" and contains a table with the following data:

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	100
<input type="checkbox"/> G.726 8 kb/s	Yes	101

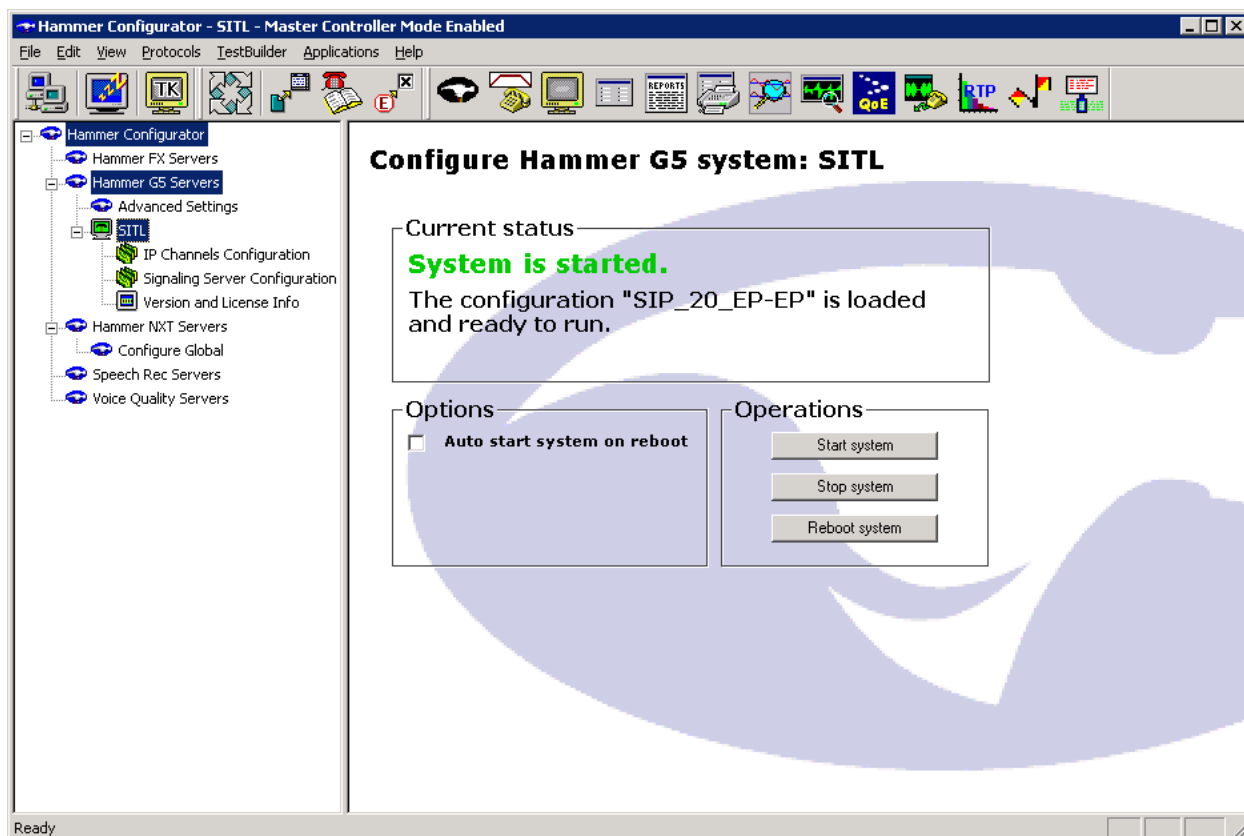
Below the table are two buttons: a blue up arrow and a blue down arrow. The "Optional Descriptions" section includes two checkboxes: "Include Field?" (unchecked) and "Field" (unchecked). The "Value" field is empty. Below this are two more checkboxes: "(i=) Media Information:" (unchecked) and "(b=) Bandwidth Information:" (unchecked). The "(a=) Attributes" section has a large empty text area and three buttons: "Add", "Edit", and "Delete". At the bottom are "OK", "Cancel", and "Help" buttons.

This completes the configuration of the SIP endpoint interface. 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.

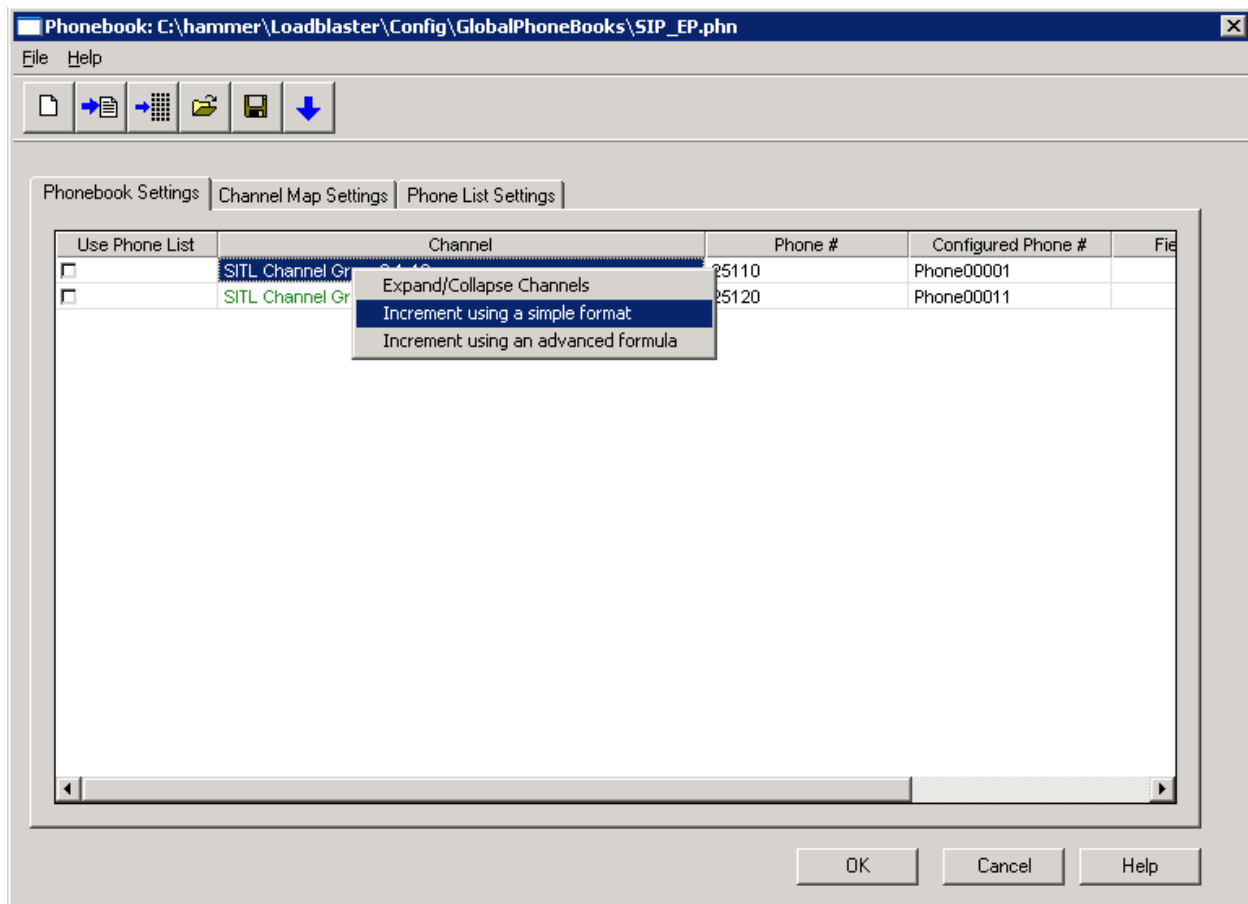


Check that the system has been started by clicking on the server name (e.g., SITL) in the left pane of the **Hammer Configurator**. If the current status is *System Is Stopped*, click the **Start system** button to start the system. When the system is started, it should appear as shown below and should also specify which configuration has been applied. The configuration performed above was saved as *SIP_20_EP-EP*. When the system is started, the Hammer G5 will register the SIP endpoints with SIP Enablement Services.



6.2.2 Configure the PhoneBook

The **PhoneBook** is used to specify which number each originating channel should dial when placing a call. Click on the **PhoneBook** icon in the **Hammer Configurator**. The **PhoneBook** window is displayed below. Since a SIP configuration has already been applied to the system, the **Channel** column is automatically displayed with the appropriate channel groups. Right-mouse click on the first line corresponding to the group of originating channels (channels 1-10) and select the **Increment using a simple format** option as shown below.



In the **Simple Incrementer** window, specify the number that the first originating channel should dial in the **Start Value** field. In this example, the first channel will dial *25110*, which corresponds to channel 11. Set the **Increment By** field to *1*. This specifies that the subsequent channels should increment the number in the **Start Value** by one. For example, channel 1 will dial 25110, channel 2 will dial 25111, channel 3 will dial 25112, and so on. The **Start Channel** field should be set to the first channel number and the **End Channel** field should be set to the last channel number. The last channel number is *20* since there are *10* originating channels and *10* terminating channels. Click **OK**.

Simple Incrementer

Server: SITL

Column: Phone #

Destination Server:

Fill Type

☒ Channel Fill ☐ Group Fill

Tel.Numbers | IP Addresses | URLs | MAC Addresses

☐ All numbers around non-numerics
(e.g. 9,,1,,888-555-999# becomes 9,,1,,888-556-*000#)

☒ xxx-yyy-zzzz

☐ xxx (e.g. 1 (888) 555-9999 becomes 1 (889) 555-9999)

☐ yyy (e.g. 1 888 555 9999 becomes 1 888 556 9999)

☒ zzzz (e.g. 1-888-555-9999 becomes 1-888-555-0000)

☐ Use H323 formatting, with prefix:

Start Value: 25110

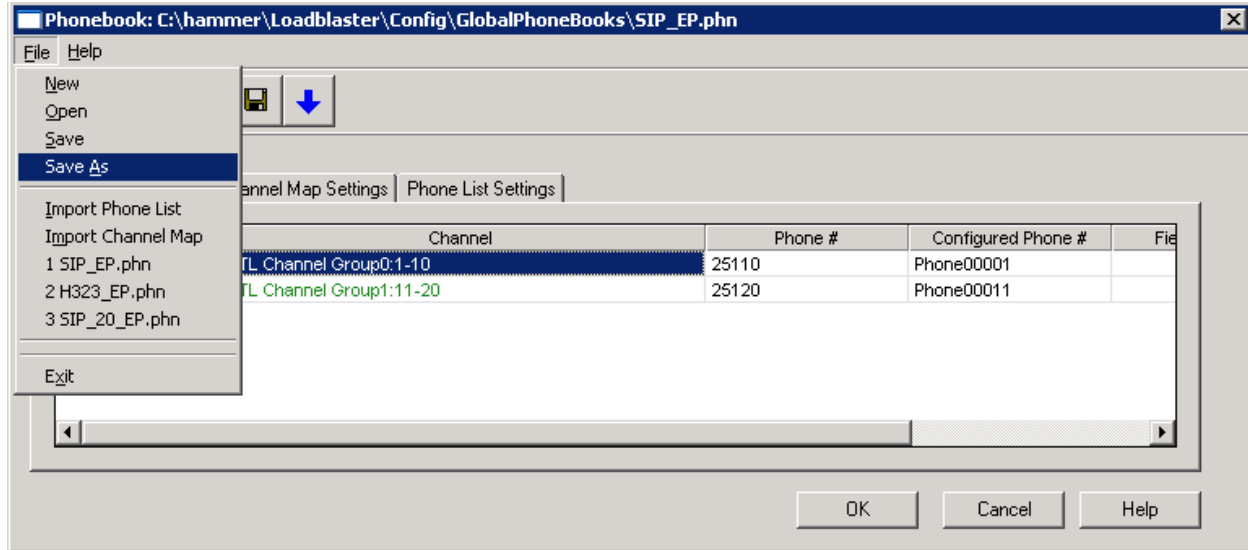
Increment By: 1

Start Channel: 1

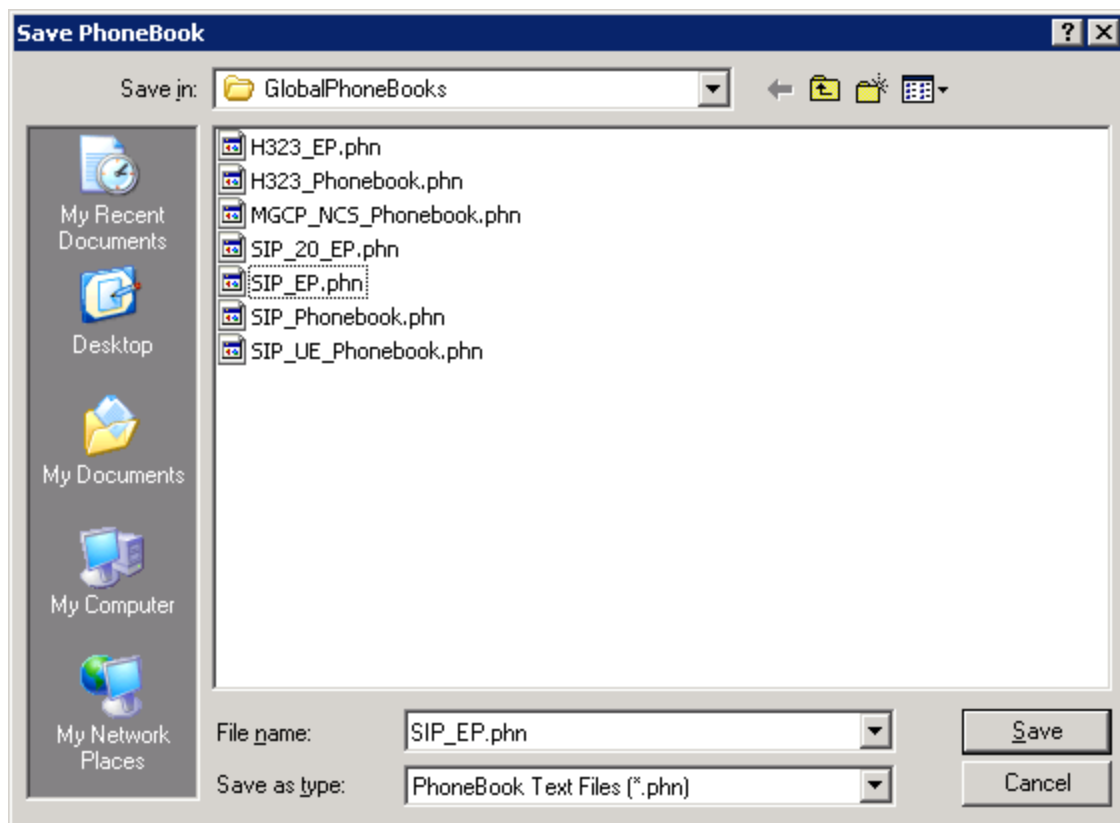
End Channel: 20

OK Cancel Apply Help

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

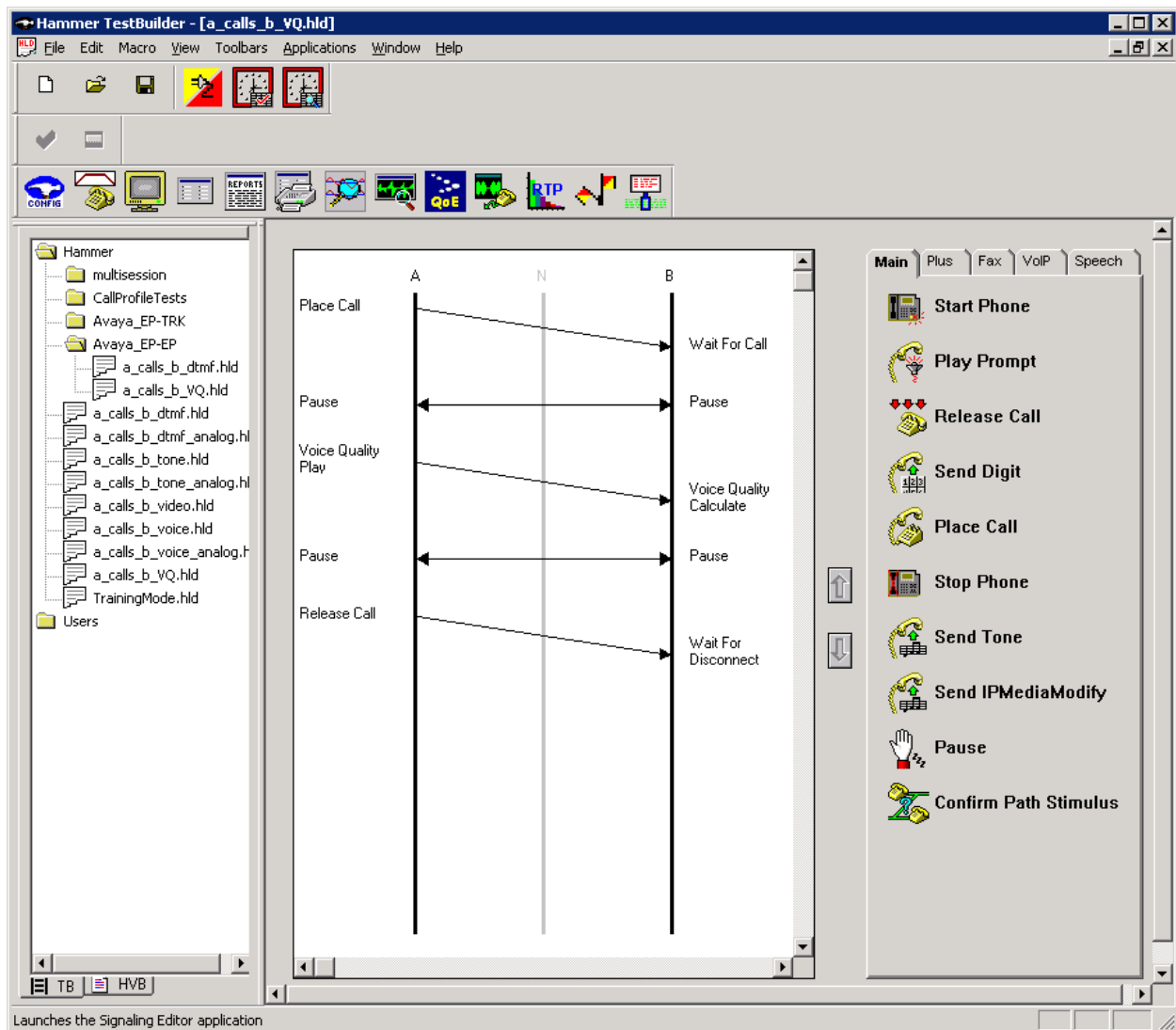


The phone book is saved as *SIP_EP.phn* in the following window. This phone book will be used when running the test.

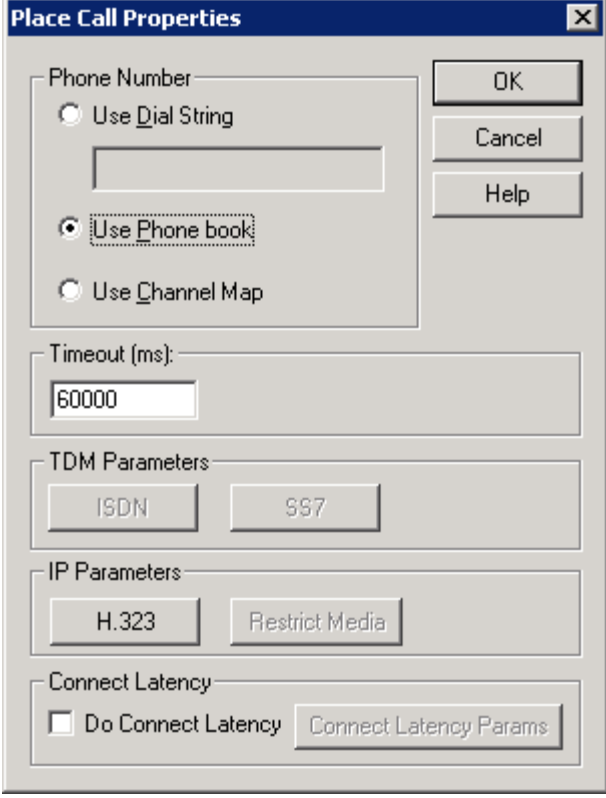


6.3 Configure and Run the Test Script

This section covers the configuration of a test script. The sample test script covered here establishes a VoIP call between two SIP endpoints on the Hammer G5, 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 is displayed in a ladder diagram shown below. Each action in the test script can be configured by double clicking on it. For detailed instructions on how to create the test script, refer to [6]. Each test step is examined below as a brief tutorial, but test scripts will vary depending on the customer's requirements.

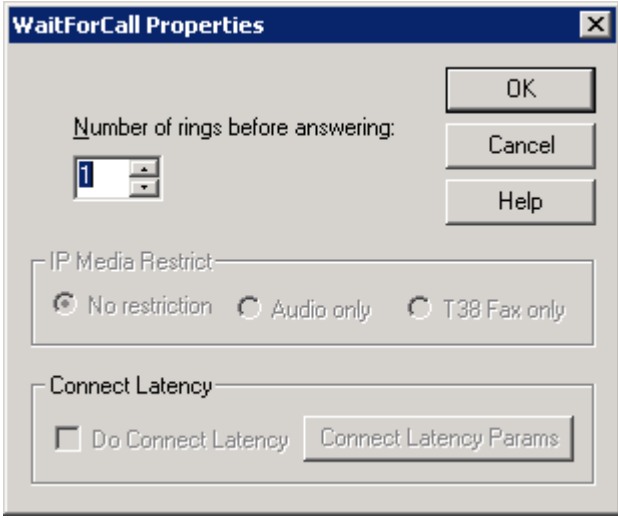


In the sample test script configured above, the A-side (originating SIP endpoint) places a call to the B-side (terminating SIP endpoint) using the **Place Call** action. The **Place Call** properties can be configured by double-clicking on the action in the ladder diagram. The **Place Call Properties**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



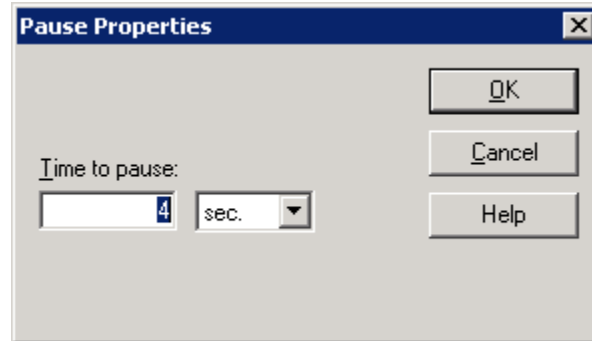
The **Place Call Properties** dialog box is shown. It has a title bar with a close button. The main area contains several sections:
1. **Phone Number**: A group box containing three radio buttons:
- **Use Dial String**: Unselected.
- **Use Phone book**: Selected.
- **Use Channel Map**: Unselected.
Below the radio buttons is an empty text input field.
2. **Timeout (ms):**: A text input field containing the value **60000**.
3. **TDM Parameters**: A group box containing two buttons: **ISDN** and **SS7**.
4. **IP Parameters**: A group box containing two buttons: **H.323** and **Restrict Media**.
5. **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, there are three buttons: **OK**, **Cancel**, and **Help**.

The B-side then waits for one ring before answering the call as configured in the **Wait for Call** action.

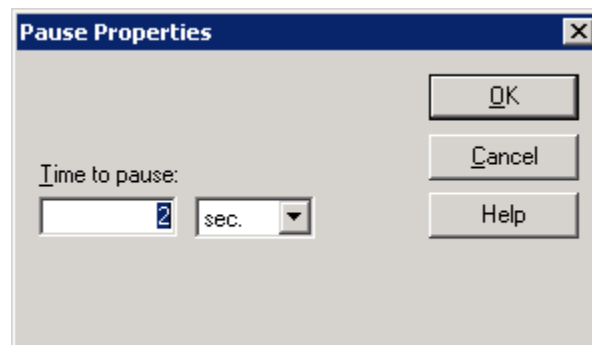


The **WaitForCall Properties** dialog box is shown. It has a title bar with a close button. The main area contains several sections:
1. **Number of rings before answering:**: A spin box showing the value **1**.
2. **IP Media Restrict**: A group box containing three radio buttons:
- **No restriction**: Selected.
- **Audio only**: Unselected.
- **T38 Fax only**: Unselected.
3. **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, there are three buttons: **OK**, **Cancel**, and **Help**.

After the call is established, the A-side pauses for 4 seconds before proceeding.

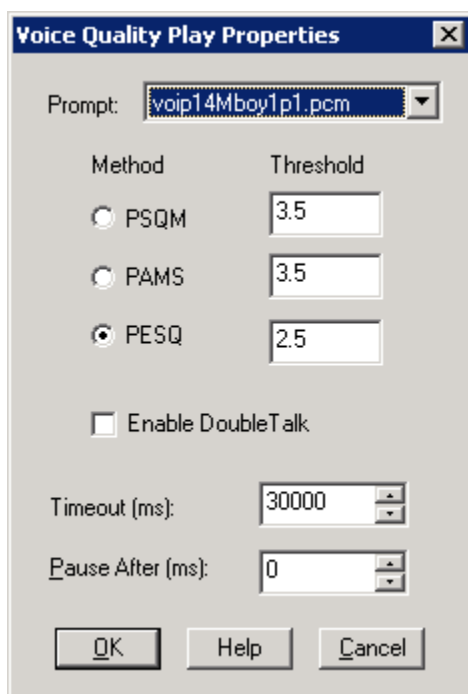


After the call is established, the B-side pauses for 2 seconds before proceeding. The B-side waits for less time because it needs to start listening for the audio prompt before the A-side starts to play it.



Important Note: In order for the call to succeed, the B-side needs to start listening for the audio prompt before the A-side starts to play it. During the compliance test, the B-side started to listen for the prompt about 2 seconds before the A-side started to play it. This can be controlled by the length of the **Pause** step described above. However, the B-side should not start listening too early (e.g., 5 seconds before the A-side starts playing the prompt or the call will most likely fail).

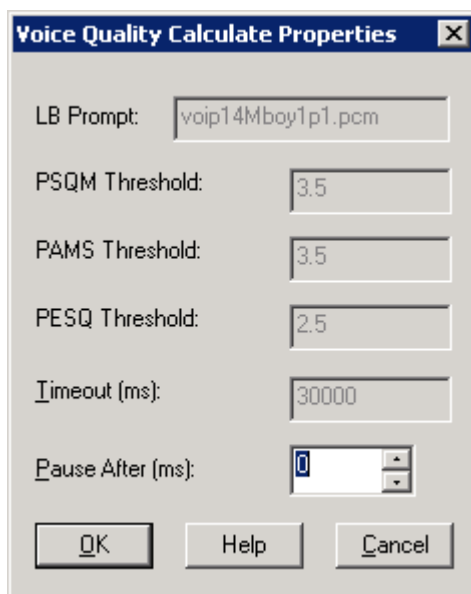
After the pause by the A-side, it starts to play the audio prompt specified in the **Prompt** field of the **Voice Quality Play Properties** window. The action properties also specify that a PESQ score should be provided.



The **Voice Quality Play Properties** dialog box contains the following fields and controls:

- Prompt:** A dropdown menu showing `voip14Mboy1p1.pcm`.
- Method** and **Threshold** section:
 - ☐ PSQM with a threshold of `3.5`
 - ☐ PAMS with a threshold of `3.5`
 - ☒ PESQ with a threshold of `2.5`
- ☐ **Enable DoubleTalk**
- Timeout (ms):** A numeric field set to `30000`.
- Pause After (ms):** A numeric field set to `0`.
- Buttons: **OK**, **Help**, and **Cancel**.

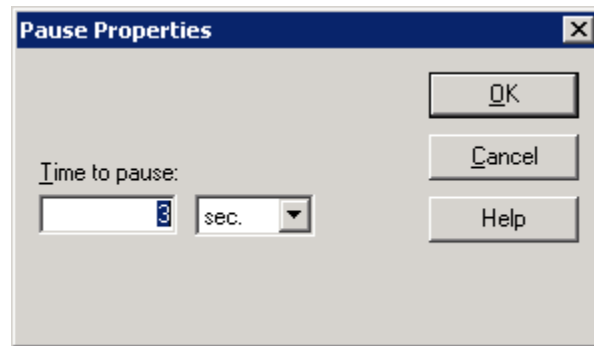
The B-side starts to listen to prompt to calculate the voice quality score.



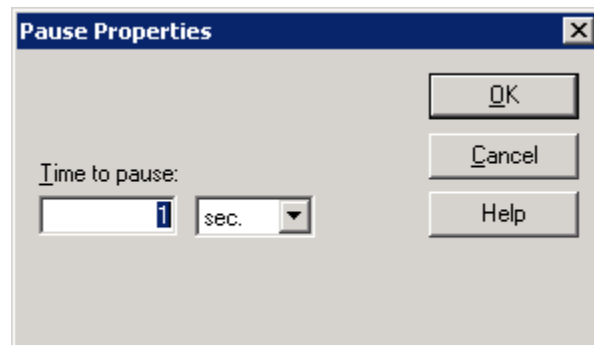
The **Voice Quality Calculate Properties** dialog box contains the following fields and controls:

- LB Prompt:** A text field containing `voip14Mboy1p1.pcm`.
- PSQM Threshold:** A numeric field set to `3.5`.
- PAMS Threshold:** A numeric field set to `3.5`.
- PESQ Threshold:** A numeric field set to `2.5`.
- Timeout (ms):** A numeric field set to `30000`.
- Pause After (ms):** A numeric field set to `0`.
- Buttons: **OK**, **Help**, and **Cancel**.

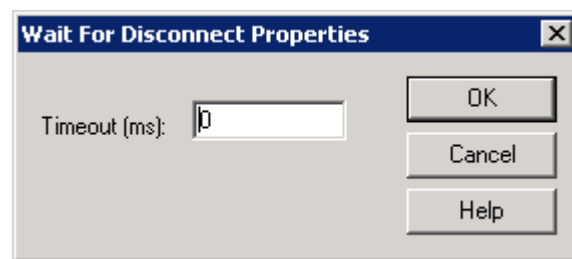
After the audio prompt is played the voice quality score is calculated, the A-side pauses for 3 seconds as shown below before releasing the call.



The B-side pauses for 1 second before releasing the call.

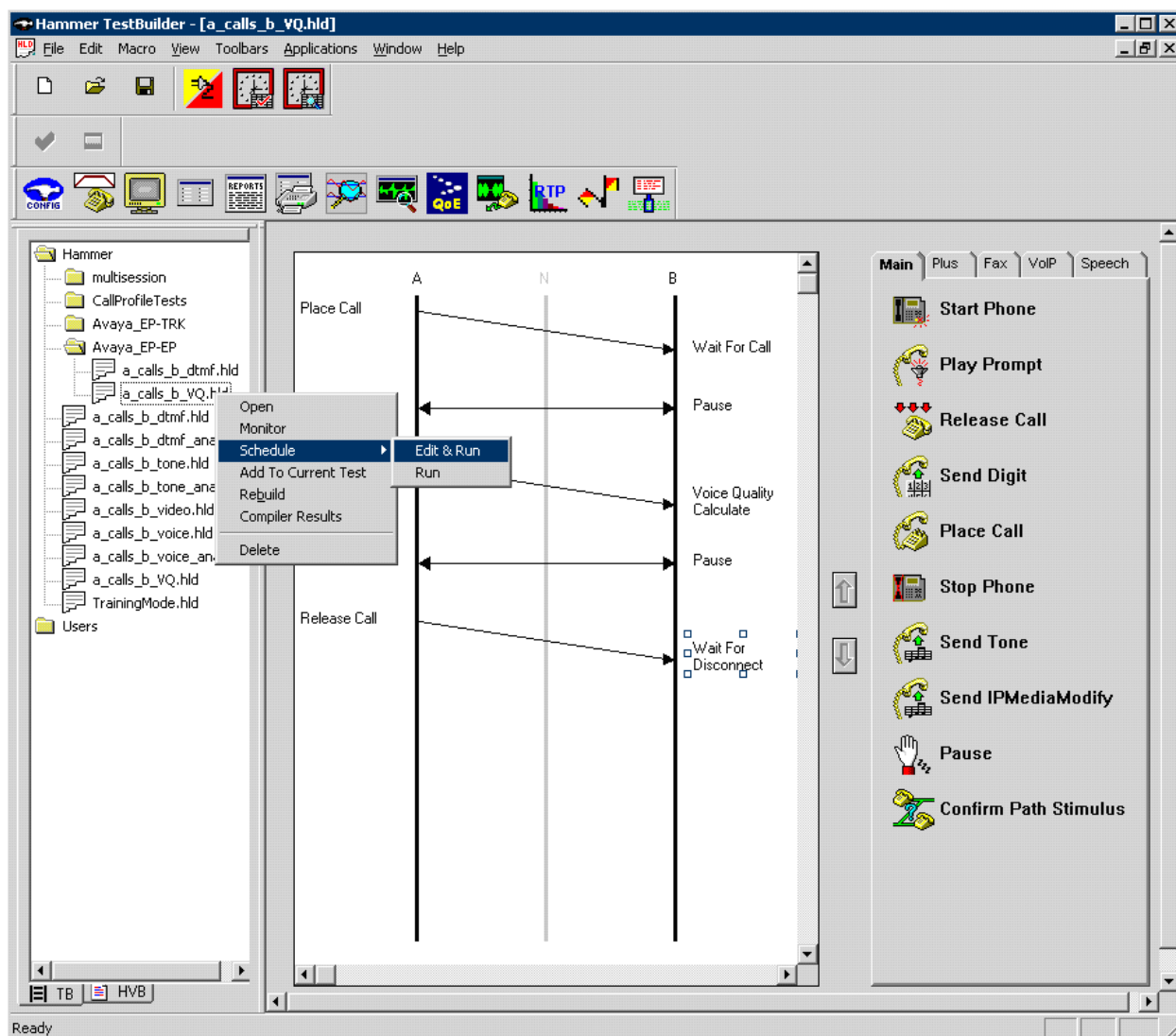


The last step of the test script is to release the call.



Once the test script is configured, save it from the **Hammer TestBuilder** application. Anytime a change is made to the test script, it needs to be saved before it can be run. In this example, the test script was saved as “a_calls_b_VQ.hld”.

To run the test, right-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, select the appropriate phone book (e.g., SIP_EP). The SIP_EP phone book was configured above. Set the **Loop Count** to the appropriate value to control the number of times 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 **Stagger** section allows the user to specify how long to wait before the test is run on the next channel. Next, click on the ellipses button (...) in the **Channels** section to assign channels to the **A-Side** and **B-Side**, which open the **Channel Browser** window.

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 G5. These parameters can slow down the test to a rate that can be reasonably handled by the test configuration.

Properties

TB Scheduler | Other

Script: ...laster\Library\Hammer\Avaya_EP-EP\A_calls_b_VQ.hld

Start Time: 2:35:17 PM 4/21/2010

Action if a Channel is busy: Wait

Channels:

A-Side: SITL[1-10]

B-Side: SITL[11-20]

PhoneBook:

Select a PhoneBook: SIP_EP

Stagger:

☐ Automatic - Est. CHT (s) 5

☐ User Defined - (ms) 0

☐ Random - Min (s) 1 Max (s) 5

☒ None

Max Active Connections: 0 (0 = Unlimited)

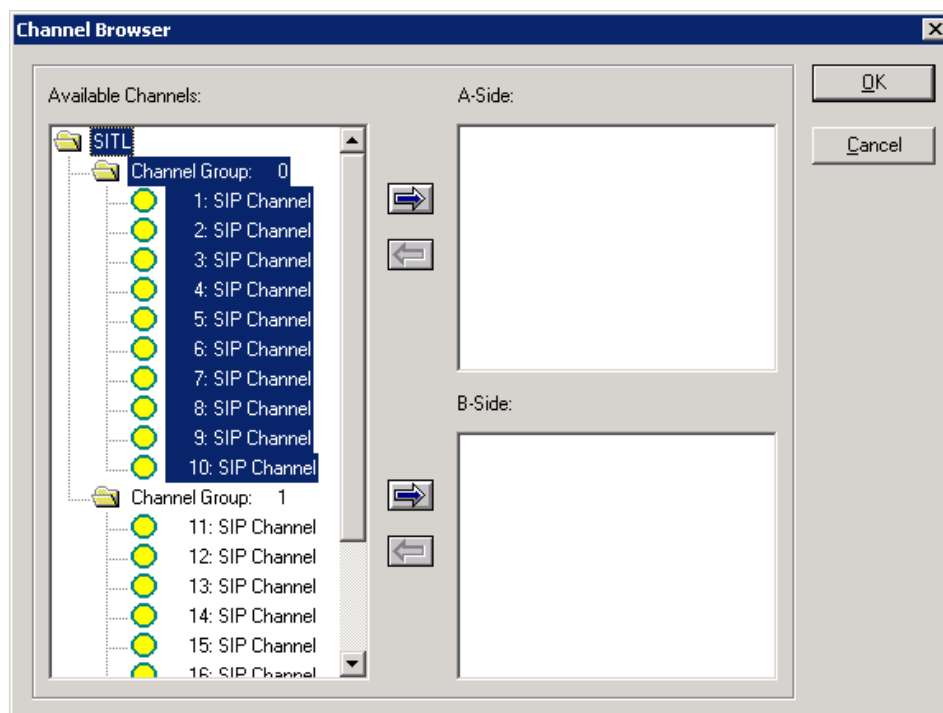
Max Test Time: Hours: 0 Minutes: 0 (0 = Forever)

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

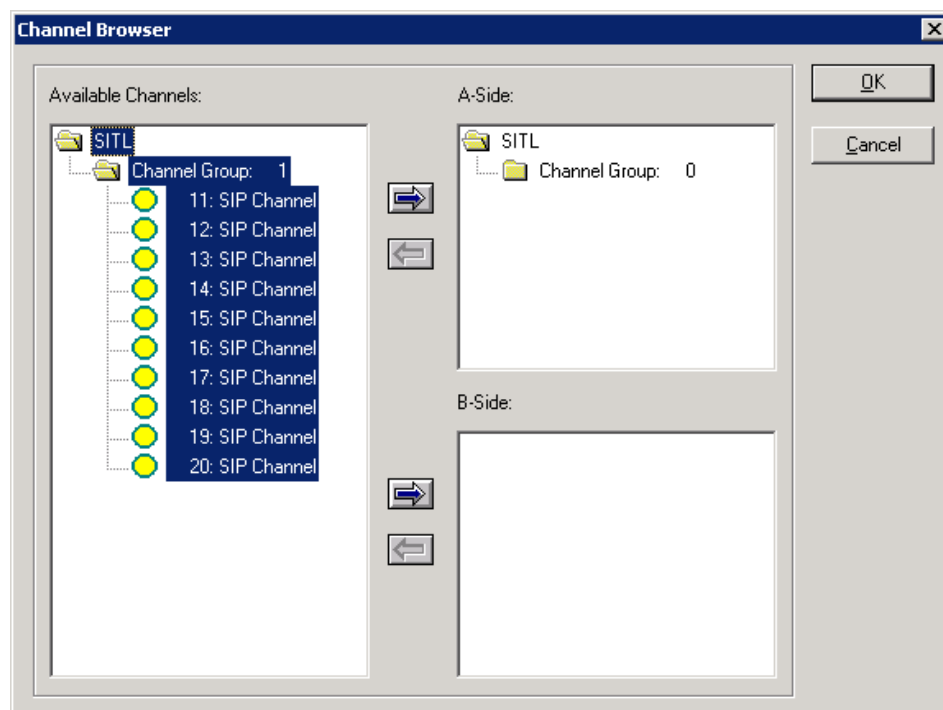
Guard Time (ms): 1500

OK Cancel Apply Help

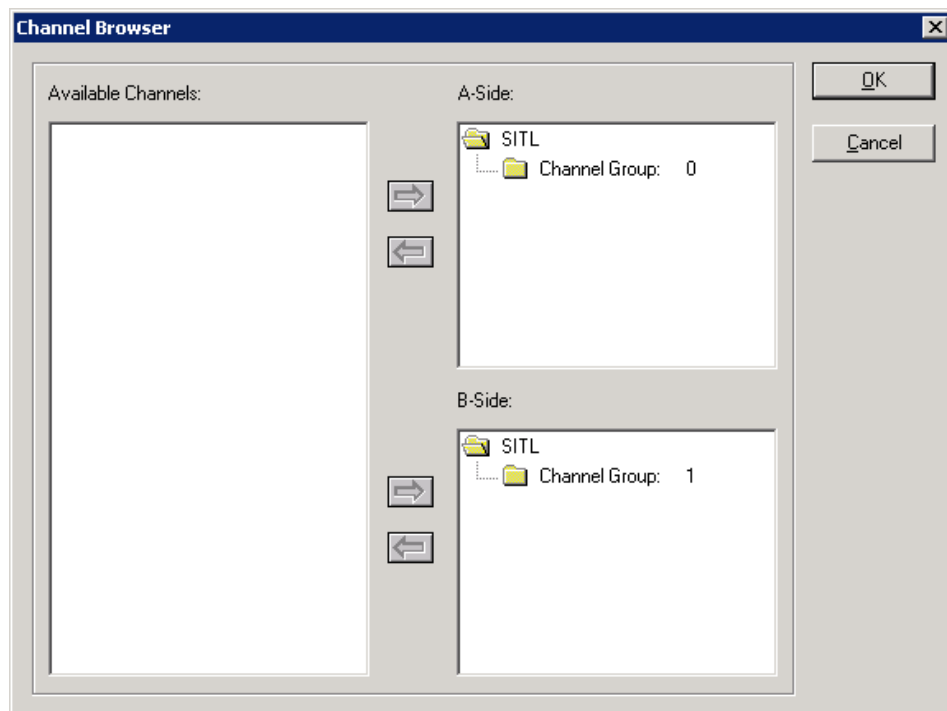
In the **Channel Browser**, select the channels in **Channel Group 0** and assign it to the **A-Side** by clicking on the appropriate right arrow.



Select the channels in **Channel Group 1** and assign it to the **B-Side** by clicking on the appropriate right arrow.




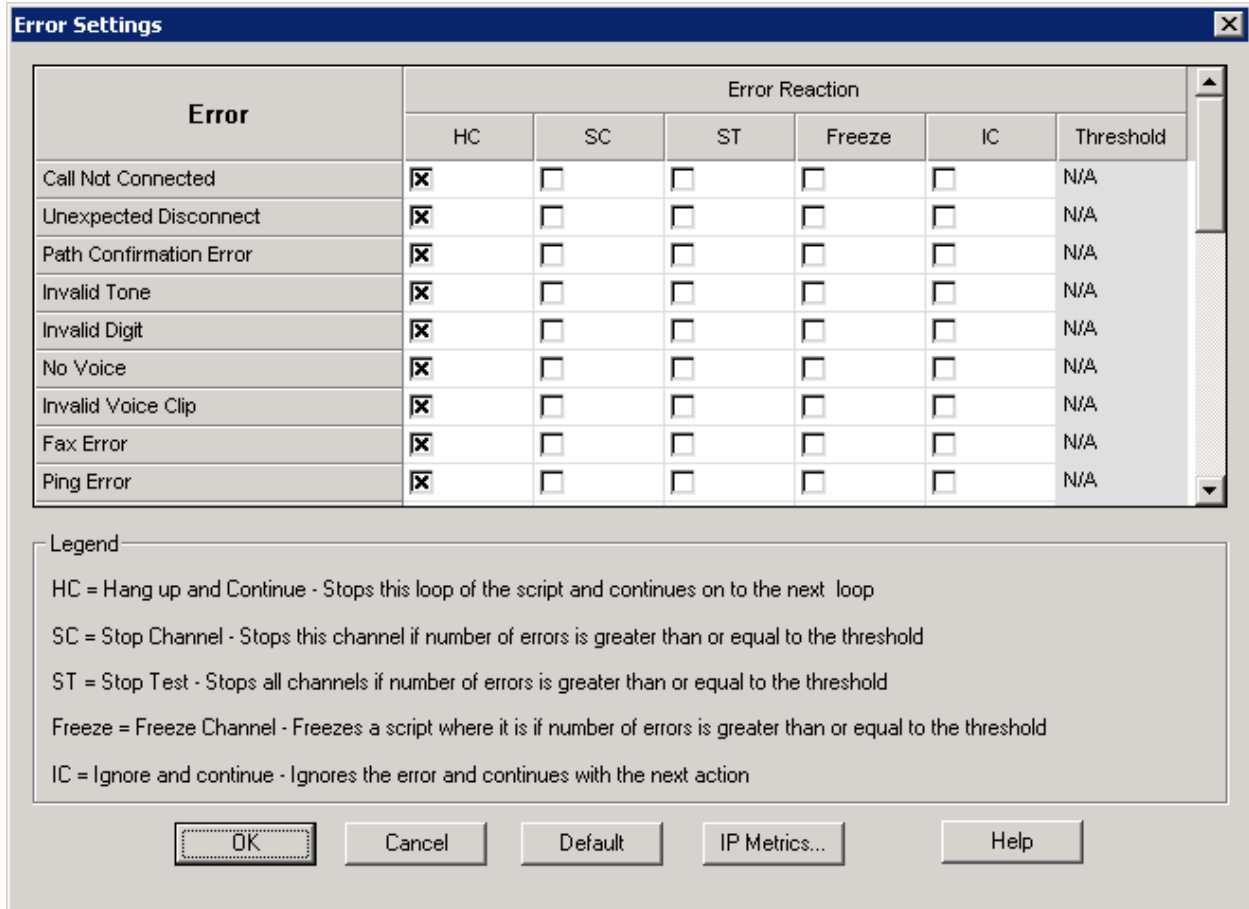
After assigning the channels to the **A-Side** and **B-Side**, the **Channel Browser** window should appear as follows. Click **OK** to return to the **Properties** window.



In the **Properties** window, click **OK** to start the test. Monitoring the test progress is covered in the **Verify Empirix Hammer G5** section (**Section 8.3**).

Helpful Tip: Once a test starts to run, if an error is encountered on a particular channel, the test will stop running on that channel unless the Hammer G5 is configured to continue running. To allow a test to continue running after an error occurs, modify the **Error Settings** by clicking on

the  icon on the **Hammer Configurator**. Click on the **HC** checkbox for each error type. This will allow the test to hang up and continue when an error occurs.



Error	Error Reaction					
	HC	SC	ST	Freeze	IC	Threshold
Call Not Connected	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	N/A
Unexpected Disconnect	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	N/A
Path Confirmation Error	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	N/A
Invalid Tone	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	N/A
Invalid Digit	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	N/A
No Voice	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	N/A
Invalid Voice Clip	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	N/A
Fax Error	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	N/A
Ping Error	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	N/A

Legend

HC = Hang up and Continue - Stops this loop of the script and continues on to the next loop

SC = Stop Channel - Stops this channel if number of errors is greater than or equal to the threshold

ST = Stop Test - Stops all channels if number of errors is greater than or equal to the threshold

Freeze = Freeze Channel - Freezes a script where it is if number of errors is greater than or equal to the threshold

IC = Ignore and continue - Ignores the error and continues with the next action

OK Cancel Default IP Metrics... Help

7 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 the SIP endpoint channels on the Hammer G5 and establishing the calls through the Avaya SIP telephony network. The interoperability compliance testing focused on the following areas:

- SIP endpoint registration with SIP Enablement Services.
- Generation of moderate call load from the Hammer G5 to Communication Manager via the SIP endpoint interface with the call being terminated on the Hammer G5.
- Support of the G.711MU, G.729AB and G.723-6.3K audio codecs.
- Voice quality as measured by PESQ scores and path confirmation.
- Support of non-direct audio and direct audio with media shuffling.
- Recovery from adverse conditions during a load test.

The compliance test also covered monitoring various reports on the Hammer G5 during and after the test runs, and checking the status of various SIP resources on Communication Manager and SIP Enablement Services.

The serviceability testing focused on verifying the ability of the Hammer G5 to recover from adverse conditions, such as disconnecting the Ethernet cable. Disconnecting the Ethernet cable would obviously cause errors in the test run, but the test would continue to run if configured to do so, otherwise the test would stop running after encountering errors.

All test cases passed. Empirix Hammer G5 was successfully compliance tested with Communication Manager and SIP Enablement Services using SIP endpoint emulation.

8 Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, SIP Enablement Services and Empirix Hammer G5.

8.1 Verify Avaya Aura™ Communication Manager

When the Hammer G5 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 SIP Enablement Services. This command should specify the trunk group and trunk member used for the call be specified.

status trunk 10/1	Page 1 of 3
TRUNK STATUS	
Trunk Group/Member: 0010/001	Service State: in-service/active
Port: T00037	Maintenance Busy? no
Signaling Group ID: 10	
IGAR Connection? no	
Connected Ports: T00082	

Page 5 of the **status trunk** command indicates the codec being used for the call and whether the call is shuffled.

status trunk 10/1	Page 2 of 3
CALL CONTROL SIGNALING	
Near-end Signaling Loc: 01A0417	
Signaling IP Address	Port
Near-end: 10.32.24.20	: 5061
Far-end: 10.32.24.100	: 5061
H.245 Near:	
H.245 Far:	
H.245 Signaling Loc:	H.245 Tunneled in Q.931? no
Audio Connection Type: ip-direct	Authentication Type: None
Near-end Audio Loc:	Codec Type: G.711MU
Audio IP Address	Port
Near-end: 10.32.24.175	: 10030
Far-end: 10.32.24.165	: 10010
Video Near:	
Video Far:	
Video Port:	
Video Near-end Codec:	Video Far-end Codec:

8.2 Verify Avaya Aura™ SIP Enablement Services

The registered SIP endpoints can also be viewed from SIP Enablement Services by navigating to **Users**→**Search Registered Users**.

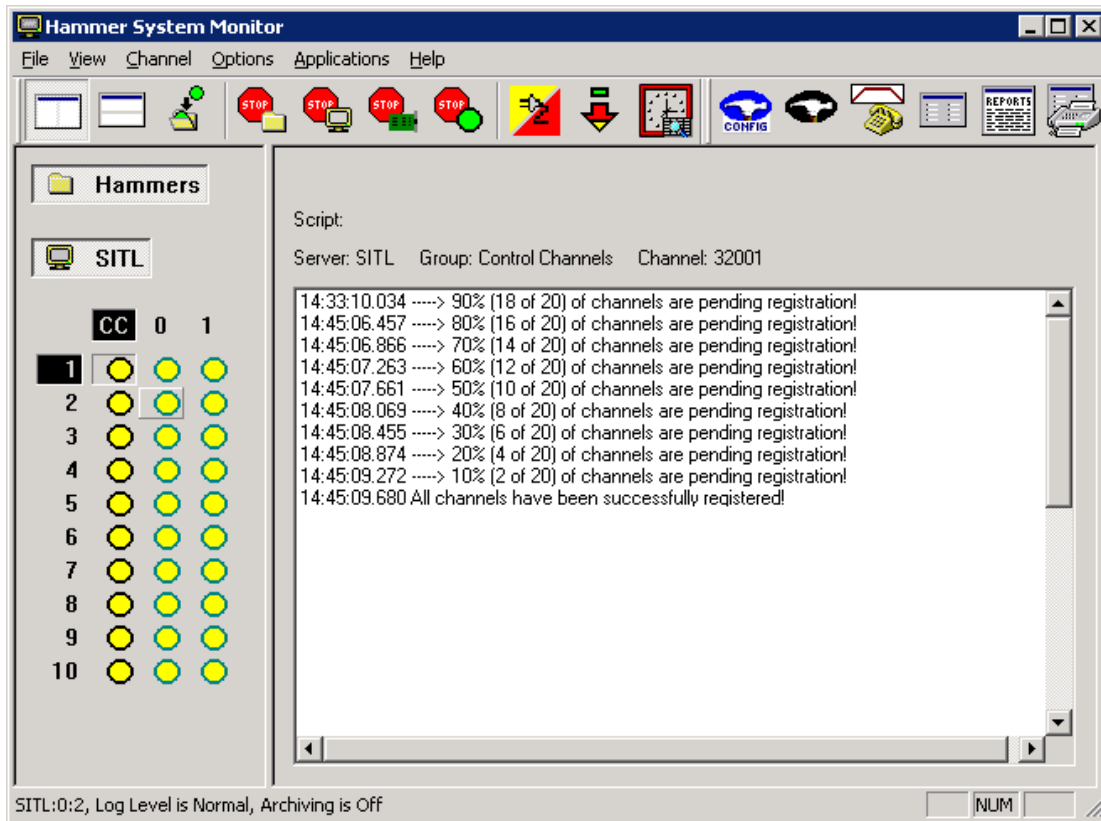
Note: Make sure that all registered SIP endpoints associated with the Hammer G5 only have one IP address.

The screenshot displays the Avaya Integrated Management SIP Server Management web interface. The top header includes the Avaya logo, the title "Integrated Management SIP Server Management", and a status indicator "This Server: [1] ses1". A navigation menu on the left lists various system management options. The main content area is titled "Registered Users on 10.32.24.100" and shows a table of registered contacts. The table has three columns: "Handle and Name", "Address", and "Expires". It lists 10 users, each with a checkbox, email address, SIP ID, and expiration time.

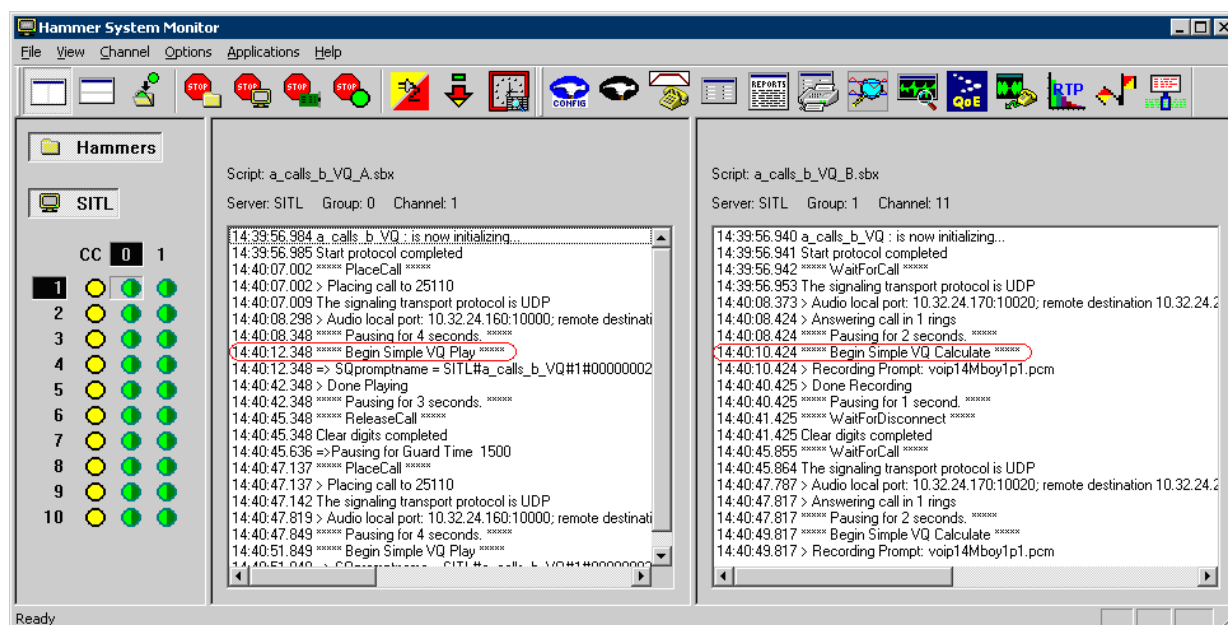
Handle and Name	Address	Expires
<input type="checkbox"/> 25100@avaya.com SIP0, Hammer	sip:25100@10.32.24.160:5060;transport=udp	Wed, 21 Apr 2010 11:02:46 EDT
<input type="checkbox"/> 25101@avaya.com SIP1, Hammer	sip:25101@10.32.24.161:5060;transport=udp	Wed, 21 Apr 2010 11:02:25 EDT
<input type="checkbox"/> 25102@avaya.com SIP2, Hammer	sip:25102@10.32.24.162:5060;transport=udp	Wed, 21 Apr 2010 11:02:13 EDT
<input type="checkbox"/> 25103@avaya.com SIP3, Hammer	sip:25103@10.32.24.163:5060;transport=udp	Wed, 21 Apr 2010 11:02:30 EDT
<input type="checkbox"/> 25104@avaya.com SIP4, Hammer	sip:25104@10.32.24.164:5060;transport=udp	Wed, 21 Apr 2010 11:02:27 EDT
<input type="checkbox"/> 25105@avaya.com SIP5, Hammer	sip:25105@10.32.24.165:5060;transport=udp	Wed, 21 Apr 2010 11:01:11 EDT
<input type="checkbox"/> 25106@avaya.com SIP6, Hammer	sip:25106@10.32.24.166:5060;transport=udp	Wed, 21 Apr 2010 11:01:35 EDT
<input type="checkbox"/> 25107@avaya.com SIP7, Hammer	sip:25107@10.32.24.167:5060;transport=udp	Wed, 21 Apr 2010 11:03:17 EDT
<input type="checkbox"/> 25108@avaya.com SIP8, Hammer	sip:25108@10.32.24.168:5060;transport=udp	Wed, 21 Apr 2010 11:03:52 EDT

8.3 Verify Empirix Hammer G5

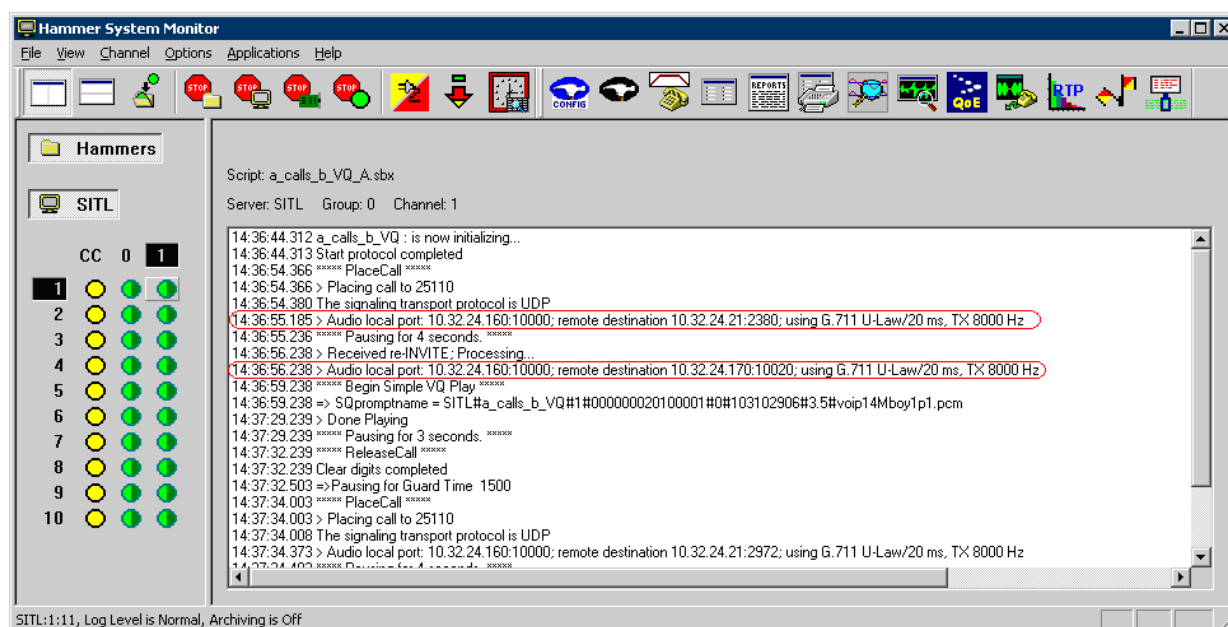
To view the SIP registration status from the Hammer G5, make sure that the **Hammer System Monitor** is running before starting the system. Click on the yellow circle under the CC column and row 1. The Hammer G5 will indicate when all of the channels have successfully registered.



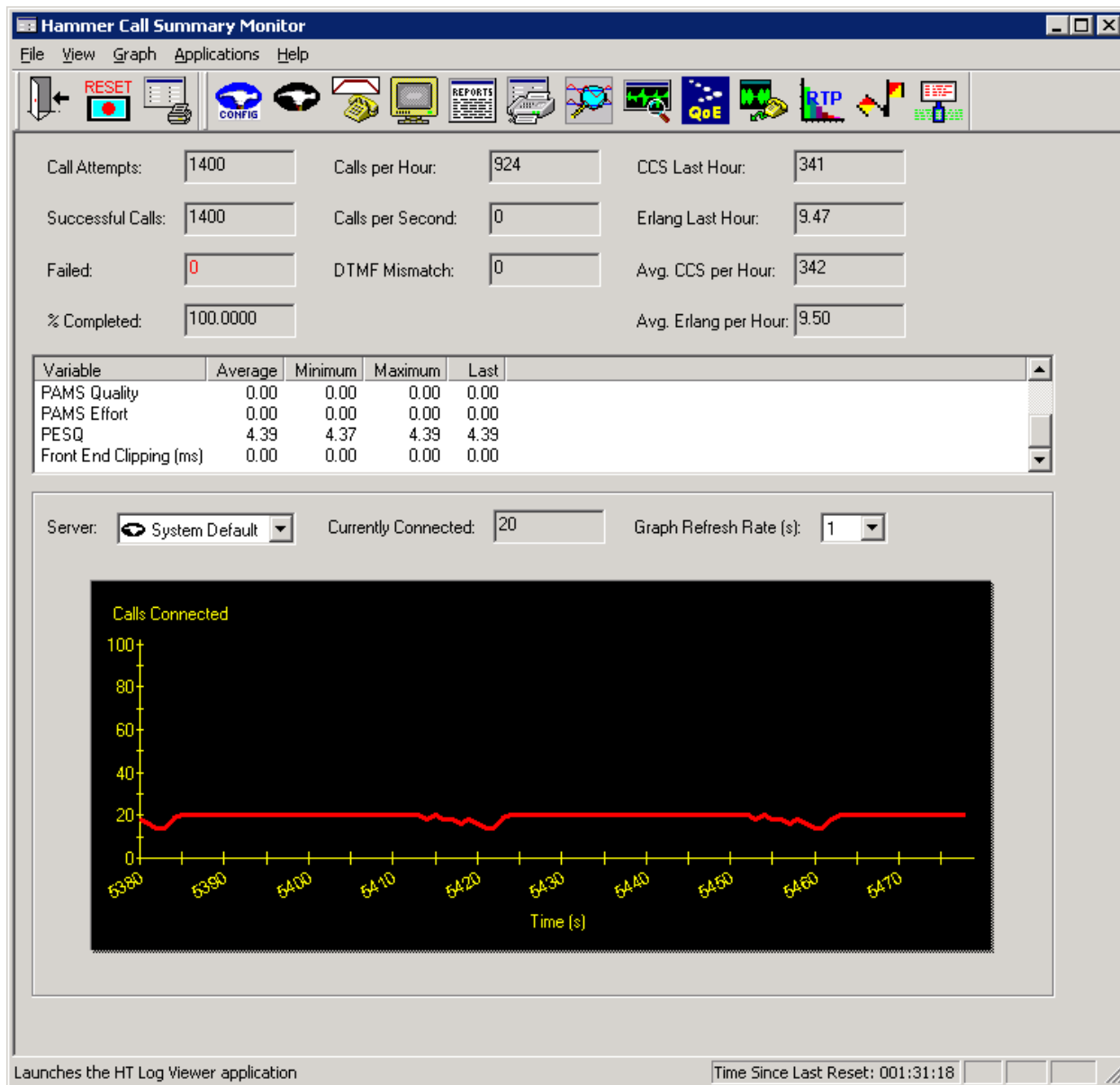
Call progress can be monitored in the **Hammer System Monitor**. The call log for an originating channel may be logged to the left window and the call log for a terminating channel may be logged to the right window. This call monitor is also useful for checking that the terminating channel is listening prior to the originating channel playing the prompt when a voice quality score is required.



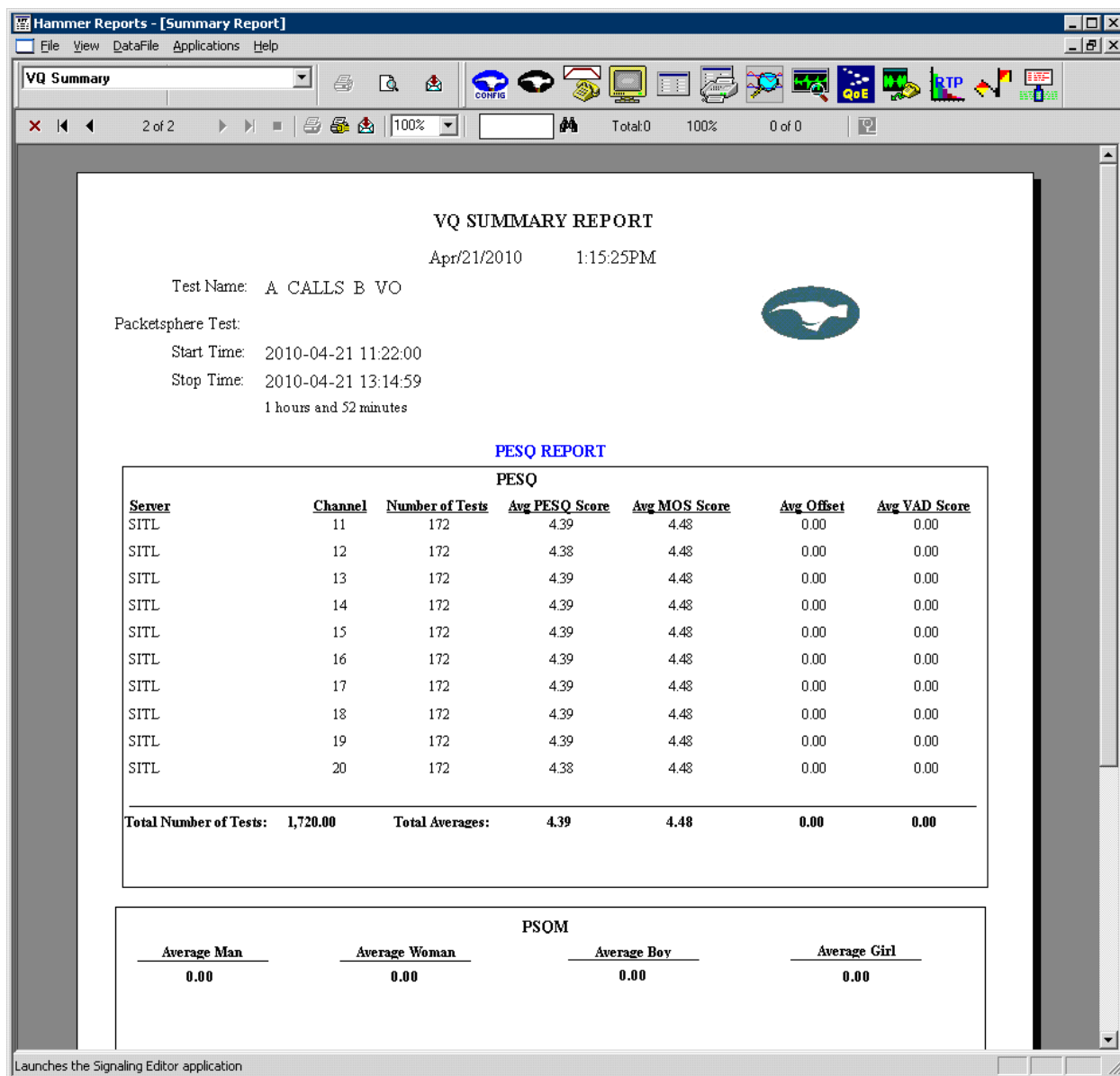
The **Hammer System Monitor** can be used to verify that a call was shuffled. This is noted by the Hammer G5 displaying the local and remote destinations in the call log.



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.



Hammer Reports can be generated by clicking on the **Reports** icon in the **Hammer Call Summary Monitor**. Various reports are available, such as the **VQ Summary** report shown below, which displays the PESQ score for the calls on each channel.



9 Conclusion

These Application Notes describe the configuration steps required to integrate the Empirix Hammer G5 with an Avaya SIP telephony network using SIP endpoint emulation. The Hammer G5 was able to register with Avaya AuraTM SIP Enablement Services, successfully establish calls through Avaya AuraTM Communication Manager, 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 AuraTM Communication Manager*, Document 03-300509, Issue 5.0, Release 5.2, May 2009, available at <http://support.avaya.com>.
- [2] *SIP Support in Avaya AuraTM Communication Manager Running on the Avaya S8xxx Servers*, May 2009, Issue 9, Document Number 555-245-206, available at <http://support.avaya.com>.
- [3] *Installing, Administering, Maintaining, and Upgrading Avaya AuraTM SIP Enablement Services*, May 2009, Issue 7, Document Number 03-600768, available at <http://support.avaya.com>.
- [4] *Application Notes for Empirix Hammer G5 with Avaya AuraTM Communication Manager Avaya AuraTM SIP Enablement Services using SIP Trunk Emulation*, Issue 1.0, available at <http://www.avaya.com>.
- [5] *Installation Guide Hammer G5TM*, Revision B, June 2009, available from Empirix.
- [6] *Getting Started Guide Hammer G5TM*, Document Number 565-005-62, Revision E, June 2009, available from Empirix.

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