

Avaya Solution & Interoperability Test Lab

Application Notes for Empirix FX-IP with Avaya Communication Manager and SIP Enablement Services using the SIP Trunk Interface – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Empirix FX-Base-IP (FX-IP) to successfully interoperate with Avaya Communication Manager using the SIP Trunk interface via Avaya SIP Enablement Services.

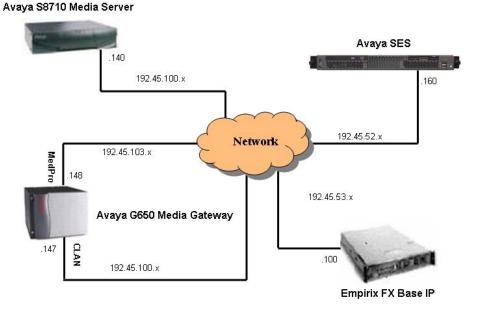
Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

Introduction

Empirix FX-IP is an integrated IP and PSTN telephony test system with Ethernet interfaces. The system generates real voice streams to simulate real-world loads, and performs real time voice quality measurements.

The FX-IP can function as a call generator or a switch. The compliance testing focused on the FX-IP as a SIP call generator, used to load Avaya Communication Manager via Avaya SIP Enablement Services (SES). In the SIP integration of FX-IP with Avaya Communication Manager, FX-IP originates and terminates calls to Avaya SES using a trunk, therefore registration was not required. To accomplish SIP Trunking between the two systems, the following was performed:

- The FX-IP generates non-registered calls over a SIP trunk direct toward Avaya Communication Manager
- Avaya Communication Manager receives the calls and redirects through AAR
- The calls are received and terminated at the FX-IP



Network Configuration Used for Compliance Test

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1.1. FX-IP Capacity

For the compliance testing, ten simultaneous calls were configured and launched on FX-IP, thus 20 channels on the Avaya Communication Manager and FX-IP were used.

The FX-IP capacities are licensing based, however the Empirix FX-IP base model has a maximum capacity of 288 channels of signaling and media or 500 channels capacity of signaling only.

2. Equipment and Software Validated

The following equipment and software were used for the configurations provided:

Equipment	Software
Avaya S8710 Media Servers	Avaya Communication Manager 4.0.0, load 730.5
 Avaya G650 Media Gateway TN799DP C-LAN Circuit Pack TN2302AP IP Media Processor Circuit Pack 	HW01 FW015 HW13 FW095
Avaya C364T-PWR Converged Stackable Switch	4.5.14
Avaya SIP Enablement Services	SES03.1.2-03.1.309.0
Empirix FX-Base-IP	2.4.1

3. Configure Avaya Communication Manager

The procedures for configuring the SIP interfaces on Avaya Communication Manager include the following areas:

- Verify Avaya Communication Manager License
- Administer IP codec set and network region
- Administer IP node names for C-LAN and SES server
- Administer IP interface and data module for C-LAN
- Administer SIP trunk group
- Administer SIP signaling group
- Administer SIP trunk group members

3.1. Verify Avaya Communication Manager License

Log into the System Access Terminal (SAT) to verify that the Avaya Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the 'display system-parameters customer-options' command to verify that there is sufficient remaining capacity for SIP Trunks on **Page 2** of the **OPTIONAL FEATURES** screen. Verify the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk calls to be launched by FX-IP, multiplied by two. This is because the scenario requires two SIP trunks per simultaneous call.

display system-parameters customer-options OPTIONAL FEATURES		Page	2 of	11
IP PORT CAPACITIES		USED		
Maximum Administered H.323 Trunks:	100	87		
Maximum Concurrently Registered IP Stations:	100	3		
Maximum Administered Remote Office Trunks:	0	0		
Maximum Concurrently Registered Remote Office Stations:	0	0		
Maximum Concurrently Registered IP eCons:	0	0		
Max Concur Registered Unauthenticated H.323 Stations:		0		
Maximum Video Capable H.323 Stations:	0	0		
Maximum Video Capable IP Softphones:	0	0		
Maximum Administered SIP Trunks:	100	10		
Maximum Number of DS1 Boards with Echo Cancellation:	0	0		
Maximum TN2501 VAL Boards:	10	0		
Maximum G250/G350/G700 VAL Sources:	0	0		
Maximum TN2602 Boards with 80 VoIP Channels:	128	0		
Maximum TN2602 Boards with 320 VoIP Channels:	128	0		
Maximum Number of Expanded Meet-me Conference Ports:	0	0		
(NOTE: You must logoff & login to effect the per	rmissi	on change	es.)	

3.2. Administer IP Codec Set and Network Region

Use the 'change ip-codec-set n' command, where 'n' is an existing codec set number that will be used for integration with FX-IP. Select an audio codec type in the **Audio Codec** field, in this case 'G.711MU'. The actual codec set number and codec type may vary. **Section 5.2.1** contains a table listing of the audio codec types that successfully interoperated between FX-IP and Avaya Communication Manager from the compliance testing. Retain the default values for the remaining fields on the screen, and submit these changes.

Note: Although G.711MU is displayed below, G.729AB and G.723-6.3k were also compliance tested successfully.

```
change ip-codec-set 2 Page

IP Codec Set

Codec Set: 7

Audio Silence Frames Packet

Codec Suppression Per Pkt Size(ms)

1: G.711MU n 2 20

2:
```

Use the 'change ip-network-region n' command, where 'n' is an existing network region number that will be used for integration with FX-IP. Enter the audio codec set number from the **IP Codec Set** screen above into the **Codec Set** field. Enable (set to 'yes') the following fields to allow for audio shuffling: **Intra-region IP-IP Direct Audio**, **Inter-region IP-IP Direct Audio**, and **IP Audio Hairpinning**. Retain the default values for the remaining fields, and submit these changes. Note that the audio shuffling feature enables the originating and terminating endpoints to exchange audio streams directly, without using media resources on the IP Media Processor (MedPro) within the Avaya G650 Media Gateway.

```
change ip-network-region 2
                                                                   Page 1 of 19
                                IP NETWORK REGION
 Region: 2
                 Authoritative Domain:
Location:
   Name:
MEDIA PARAMETERS
                                 Intra-region IP-IP Direct Audio: yes
                               Inter-region IP-IP Direct Audio: yes
     Codec Set: 2
  UDP Port Min: 2048
                                            IP Audio Hairpinning? y
  UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
                                          RTCP Reporting Enabled? y
Call Control PHB Value: 34
Audio PHB Value: 46
RTCP MONITOR SERVER PARAMETERS
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
```

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3.3. Administer IP Node Names for C-LAN and SES Server

Use the 'change node-names ip' command, and add entries for the C-LAN and Avaya SES server. In this case, 'CLAN-1A06' and '192.45.100.147' are entered as **Name** and **IP Address** for the C-LAN, and 'SES' and '192.45.52.160' are entered as **Name** and **IP Address** for the Avaya SES server. The actual node names and IP addresses may vary. Submit these changes.

```
Page
                                                                      1 of
                                                                             1
change node-names ip
                                 IP NODE NAMES
   Name
                     IP Address
                                          Name
                                                            IP Address
CLAN-1A06
                   192.45 .100.147
                                                             . . .
SES
                   192.45 .52 .160
( 14 of 23 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
```

3.4. Administer IP Interface and Data Module for C-LAN

Add the C-LAN to the system configuration using the 'add ip-interface 1a06' command. Note that the actual slot number may vary. In this case, '1a06' is used as the slot number. Enter the C-LAN node name assigned from **Section 3.3** into the **Node Name** field, and then the **IP Address** will be populated automatically.

Enter proper values for the **Subnet Mask** and **Gateway Address** fields. In this case, '255.255.255.0' and '192.45.100.1' are used to correspond to the network configuration in these Application Notes. Set the **Enable Ethernet Port** field to 'y', and the **Network Region** field to the network region number from **Section 3.2**. Default values may be used in the remaining fields. Submit these changes.

```
change ip-interface 01a06
                                                                       1 of
                                                                             1
                                                                Page
                                  IP INTERFACES
                  Type: C-LAN
                  Slot: 01A06
          Code/Suffix: TN799 D
            Node Name: CLAN-1A06
           IP Address: 192.45 .100.147
          Subnet Mask: 255.255.255.0
                                                                 Link: 3
      Gateway Address: 192.45 .100.1
 Enable Ethernet Port? y
                                                Allow H.323 Endpoints? y
       Network Region: 2
                                                 Allow H.248 Gateways? y
                                                  Gatekeeper Priority: 5
                 VLAN: n
Target socket load and Warning level: 400
      Receive Buffer TCP Window Size: 8320
                                ETHERNET OPTIONS
                  Auto? y
```

Next, add a new data module using the 'add data-module n' command, where 'n' is an available extension. Enter the following values, and submit these changes.

- Name: A descriptive name.
- **Type:** 'ethernet'
- **Port:** Same slot number from the **IP INTERFACES** screen above and port '17'.
- Link: An available link number.

```
add data-module 2003

DATA MODULE

Data Extension: 2003

Type: ethernet

Port: 01A0617

Link: 3

Network uses 1's for Broadcast Addresses? y
```

3.5. Administer SIP Trunk Group

Administer a SIP trunk group to interface with the FX-IP. Use the 'add trunk-group n' command, where 'n' is an available trunk group number. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- Group Type: 'sip'
- Group Name: A descriptive name.
- TAC: An available trunk access code.
- Service Type: 'tie'

```
add trunk-group 10
                                                              1 of 21
                                                        Page
                             TRUNK GROUP
Group Number:
                                                        CDR Reports: y
                                Group Type: sip
                                                  TN: 1 TAC: 199
 Group Name: SIP Trunks to SES-DevCon1 COR: 1
  Direction: two-way Outgoing Display? n
Dial Access? n
                                              Night Service:
Queue Length: 0
                                 Auth Code? n
Service Type: tie
                                                 Signaling Group:
                                                 Number of Members: 0
```

3.6. Administer SIP Signaling Group

Administer a SIP signaling group for the newly added trunk group to use for signaling. Use the 'add signaling-group n' command, where 'n' is an available signaling group number. Enter the following values for the specified fields, and retain the default values for all remaining fields.

- Group Type: 'sip'
- Transport Method: 'tls'
- Near-end Node Name: C-LAN node name from Section 3.3.
- Far-end Node Name: SES server node name from Section 3.3.
- Far-end Listen Port: '5061'
- Far-end Network Region: Network region number from Section 3.2.
- Far-end Domain: SIP domain name of SES server from Section 4.1.

```
add signaling-group 10
                                                            Page
                                                                   1 of
                                                                          1
                               SIGNALING GROUP
Group Number: 10
                             Group Type: sip
                       Transport Method: tls
  Near-end Node Name: CLAN-1A06
                                            Far-end Node Name: SES
Near-end Listen Port: 5061
                                          Far-end Listen Port: 5061
                                      Far-end Network Region: 2
      Far-end Domain: devconnect.com
                                            Bypass If IP Threshold Exceeded? n
        DTMF over IP: rtp-payload
                                             Direct IP-IP Audio Connections? n
                                                       IP Audio Hairpinning? n
Enable Layer 3 Test? n
Session Establishment Timer(min): 120
```

3.7. Administer SIP Trunk Group Members

Use the 'change trunk-group n' command, where 'n' is the trunk group number added in Section **3.5**. Enter the signaling group number from Section **3.6** into the Signaling Group field. Enter the desired number of trunk group members into the Number of Members field, which would be the same as the desired number of simultaneous SIP calls multiplied by two. For the compliance testing, a total of ten trunk group members are administered to support five simultaneous SIP calls. Submit these changes.

```
1 of 21
change trunk-group 10
                                                           Page
                             TRUNK GROUP
                                Group Type: sip
Group Number: 10
                                                        CDR Reports: y
 Group Name: SIP Trunks to SES-DevCon1 COR: 1
                                                  TN: 1 TAC: 199
  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:
                                                                  250
```

Solution & Interoperability Test Lab Application Notes ©2007 Avaya Inc. All Rights Reserved. Add trunk members to the trunk group as displayed below. **Note:** The 'Number of Members' may vary.

```
change trunk-group 10
                                                                  5 of 21
                                                            Page
                              TRUNK GROUP
                                  Administered Members (min/max): 1/250
GROUP MEMBER ASSIGNMENTS
                                       Total Administered Members: 250
      Port
                    Name
 1: T00046
                    SIP Trunks
 2: T00047
                    SIP Trunks
                    SIP Trunks
 3: T00048
                     SIP Trunks
 4: T00049
 5: T00050
                      SIP Trunks
 6: T00051
                      SIP Trunks
 7: T00052
                      SIP Trunks
 8: T00053
                     SIP Trunks
 9: T00054
                     SIP Trunks
10: T00055
                    SIP Trunks
11: T00056
                    SIP Trunks
12: T00057
                    SIP Trunks
13: T00058
                    SIP Trunks
14: T00059
                     SIP Trunks
15: T00060
                   SIP Trunks
```

3.8. Administer the Dial Plan Information

The next three screens deal with routing the incoming call to the appropriate outbound trunk. The dial plan used for SIP trunking dealt with a 5 digit dial plan. The incoming 5 digit number was preceded by an '8'. Thus '8110xx' was the number called by the FX-IP (Note: xx is any number between '01' through '99'). The '8' was stripped off by SES, as seen in Section 4.5, and the 5 digits that remained began with '110'. Below, the 'change dialplan analysis' SAT command displays the 'Dialed String' as 110 being a 'Total Length' of '5', and the 'Call Type' is an 'ext' (extension).

change dialplan	analysis	DIAL PLAN	ληγιλά	י דאסו ד		Page 1	. of	12
		DIAL FLAN	ANALISIS	, IADUE	Perc	ent Full	1:	2
Dialed String 0 1 110 8 9	3 d. 5 e: 1 f.		Total Length 3 3		Dialed String	Total Length		

The 'change aar analysis 1' SAT command further develops the '110' dial string by associating it with the 'Route Pattern' of '10' and the 'Call Type' of 'aar'.

change aar analysis 1	л	דת סג	GIT ANALYS	נדמ האסו	Ē	Page 1 of 2	
	П	AR DI	GII ANADI.	JIJ IADI	-10	Percent Full: 2	
Dialed String 110 2	Tot Min 5 7		Route Pattern 10 999	Call Type aar aar	Node Num	ANI Reqd n n	

Lastly the 'change route-pattern 10' SAT command indicates the 'Grp No' (or Trunk Group) that will be used to route the out bound traffic, and in this case '10'.

```
change route-pattern 10
                                                        Page
                                                              1 of
                                                                     3
                Pattern Number: 10 Pattern Name: SES SIP
                         SCCAN? n Secure SIP? n
   Grp FRL NPA Pfx Hop Toll No. Inserted
                                                              DCS/ IXC
   No Mrk Lmt List Del Digits
                                                              OSIG
                        Dgts
                                                              Intw
1: 10 0
                                                               n user
2:
                                                               n user
3:
                                                               n user
4:
                                                               n user
5:
                                                               n user
6:
                                                               n user
                          ITC BCIE Service/Feature PARM No. Numbering LAR
    BCC VALUE TSC CA-TSC
   0 1 2 M 4 W Request
                                                    Dgts Format
                                                  Subaddress
1: yyyyyn n
                          rest
                                                                  none
2: yyyyyn n
                          rest
                                                                  none
3: yyyyyn n
                          rest
                                                                  none
4: yyyyyn n
                          rest
                                                                  none
5: ууууул п
                          rest
                                                                  none
                          rest
6: ууууул п
                                                                  none
```

4. Configure Avaya SIP Enablement Services

This section provides the procedures for configuring Avaya SIP Enablement Services (SES). The procedures include the following areas:

- Obtain SIP domain and host
- Administer media server
- Administer users
- Administer media server address map
- Administer trusted host

4.1. Obtain SIP Domain and Host

Access the Avaya SES administration web interface by using the URL 'http://ip-address/admin' in an Internet browser window, where 'ip-address' is the IP address of the SES server. Note that the IP address for the Avaya SES server may vary, and in this case '192.45.52.160' is used, as administered in Section 3.3. Log in with the appropriate credentials and select the Launch Administration Web Interface option.

AMAYA	R			Integrated Management Standard Management Solutions					
Help Log Off									
	-	Administration	The administration web interface allows you to administer this Converged Communication Server.	Launch Administration Web Interface					
		Maintenance	The Maintenance Web Interface allows you to maintain, troubleshoot, and configure the server.	<u>Launch Maintenance Web</u> Interface					

The **Top** screen is displayed next. If this is the initial setup of the Avaya SES server, then follow the Avaya SIP Enablement Services Installation and Administration Guide [2] to administer the SIP domain and host. These Application Notes assume the Avaya SES server has already been configured with the proper domain and host information.

Note: After adding new entries or editing exiting entries, an 'Update' will appear at the bottom of the Left Panel. The 'Update' should be performed to preserve the change(s) in the administration.

Select Server Configuration > System Properties from the Left Panel to display the Edit System Properties screen below.

Help Exit			Server: 192.45.52.160
Top ■ Users	Edit System Pr	operties	
 Conferences Media Server Extensions Emergency Contacts Hosts 	SES Version System Configuration Host Type	SES-3.1.2.0-309.0 simplex home/edge	
 Media Servers Adjunct Systems Services Server Configuration System Properties Admin Accounts License 	for a DNS domain of eastco domain would likely be con	is field, most often the SIP level DNS domain. For example, oast.example.com, the SIP figured to example.com. This t messages to users with handles	
IM Log Settings SNMP Configuration Certificate Management Generate Web Certificate Signing Request Install Web Certificate Generate SIP	Local Name	Iocalhost 192.45.52.160 SES-DevCon1.devconnect.com 192.45.52.160	
Certificate Signing Request Install SIP Certificate IM Logs Trace Logger Configure Filters Trace Manager Logs		SES-DevCon1.devconnect.com 192.45.52.1 SAMP ed.	-

Select **Hosts** from the Left Panel to display the **List Hosts** screen. For the compliance testing, only one host is administered as both the edge and home server, as indicated by the 'home/edge' value in the **Host Type** field shown below.

AVAYA						1	Integrated Management SIP Server Management
Help Exit							Server: 192.45.52.160
Top Users Conferences	List Host	s					
 Media Server Extensions 	Status	Comm	ands		Host	Туре	
Emergency Contacts	up to date Edit	Map Go-To	Test-Link	Delete	192.45.52.160	home/edge	
 Hosts List 	Force All Migrate Home/Edg						
Migrate Home/Edge	rigrate numericag						
Media Servers							
Adjunct Systems							
Services							
Server Configuration							

In the **Edit Host** screen, check the value of the **Host Type** field. Perform this check for all hosts listed in the previous **List Hosts** screen, and make a note of the name of the host that is serving as the home server, to be used later to configure the media server interface.

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AVAYA	Integrated Management SIP Server Management
Help Exit	Server: 192.45.52.160
Top • Users • Conferences	Host IP
 Media Server Extensions Emergency Contacts Hosts List Migrate Home/Edge Media Servers Adjunct Systems 	Host IP 192.45.52.160 Address* ******* DB Password ******* Profile Service ******* Password ******* Host Type home/edge Parent none Listen Protocols UDP TCP
Services Server Configuration Certificate Management IM Logs Trace Logger Configure Filters Trace Manager	Link Protocols © UDP © TCP © TLS Presence Access Policy © Allow All © Deny All (Default) Emergency Contacts Policy © Allow © Deny Minimum Registration 300 Registration Expiration Timer (seconds)* 3600 (seconds)
Logs Export/Import to ProVision	Line Reservation Timer (seconds) 30 * Outbound Pouting Allowed Internal Internal

4.2. Administer Media Server

Select **Media Servers > Add** from the Left Panel to display the **Add Media Server** screen. This screen associates a media server with a SIP domain and host. Enter the following values for the specified fields, and retain the default values for the remaining fields. Click on **Add** in the lower Right Panel at the end to submit these changes.

- Media Server Interface: A descriptive name.
- Host: Select the host name of the home server from Section 4.1.
- **SIP Trunk IP Address:** Enter the C-LAN IP address from **Section 3.3**.

The 'SIP Trunk IP Address' field is populated with the IP address of the CLAN board.

Αναγα		Integrated Management SIP Server Management
Help Exit		Server: 192.45.52.160
Top = Users List	Add Media Server	r Interface
Add Search Edit	Media Server Interface Name* Host	S8710 192.45.52.160 •
Eur Delete Password Default Profile	SIP Trunk SIP Trunk Link Type SIP Trunk IP Address*	C TCP
Registered Users Conferences Media Server Extensions Emergency Contacts Hosts	Media Server Media Server Admin Address (see Help) Media Server Admin Login	
 Media Servers List Add Add Adjunct Systems 	Media Server Admin Password Media Server Admin Password Confirm Fields marked * are required.	
Services Server Configuration System Properties Admin Accounts License		
IM Log Settings SNMP Configuration		

4.3. Administer Users

Users are not required for compliance testing SIP Trunks from the FX-IP to the Avaya Communication Manager.

4.4. Administer Media Server Address Map

Select **Media Servers** from the Left Panel to display the **List Media Servers** screen below. Click on the **Map** link associated with the media server interface administered from **Section 4.2**, in this case 'S8710'.

AVAYA Integrated SIP Serve							
Help Exit							9
Top Users Conferences	- 1	.ist Media	a Sei	vers			
Media Server Extensions		<u>Co</u>	mman	<u>ds</u>		Interface	<u>Host</u>
Emergency Contacts	Edit	Extensions	Мар	Test-Link	Delete	G350	192.45.52.160
Hosts	Edit	Extensions	Мар	Test-Link	Delete	S8710	192.45.52.160
Media Servers List Add	Add Ar	nother Media	Serve	r Interface			

On the List Media Server Address Map screen, click the Add Map In New Group link in the lower Right Panel.



The Add Media Server Address Map screen is displayed next. This screen is used to specify which calls to be routed to the media server appearing in the Host field.

For the Name field, enter a descriptive name to identify the routing.

For the **Pattern** field, enter an appropriate syntax for address mapping. The syntax in the **Pattern** field is compared to the Uniform Resource Identifier (URI) of an incoming SIP INVITE message. If a match occurs, then the call is routed to the media server. For the compliance testing, the value '^sip:8110.*' was used and explained as follows:

- **^sip:** Match to any SIP INVITE message with 'sip:' at the beginning of the URI.
- **8110** Matching the inbound extension digits beginning with '8110'.
- .* Match the last two digits of the extension to any digits.

Therefore, the pattern '^sip:8110.*' will match the extension range of 811000 - 811099. The actual **Name** and **Pattern** values may vary. The compliance testing could have different methods of pattern matching but chose the pattern depicted here for simplicity. For additional information on the pattern matching, refer to the SIP Enablement Services Installation and Administration Guide [2].

Verify the **Replace URI** field is checked, to enable SES to replace the URI in the incoming SIP INVITE messages with C-LAN contact information, in order to reach Avaya Communication Manager. Click **Add** at the bottom of the screen.

Αναγα	Integrated Management SIP Server Management
Help Exit	Server: 192.45.52.160
Top Users Conferences Media Server Extensions Emergency Contacts Hosts List Migrate Home/Edge Media Servers List Add	Host S8710 Name* FXIPInbound Pattern* ^sip:8110.* Replace URI I Fields marked * are required.

The List Media Server Address Map screen is displayed, with the Contact information automatically populated by the Avaya SES server. Note the C-LAN IP address appearing in the value of the Contact field, to enable incoming SIP messages to be forwarded to the C-LAN on Avaya Communication Manager.

Αναγα				I	ntegrated Manage SIP Server Manage	
Help Exit					Server: 192.4	
Top Users Conferences Media Server Extensions		e dia Serve 88710	er Address	s Мар		
Emergency Contacts Hosts	<u>Commands</u>	Name	<u>Commands</u>	Conta	<u>ct</u>	
Update All List	<u>Edit</u> Delete F	XIPInbound	Edit Delete	sip:\$(user @192.45.	') 100.147:5061;transport=tls	
Migrate Home/Edge	Add Another M	ар	Add Another	Contact		Delete Group
Media Servers List Add	Add Map In Net	w Group				

4.5. Administer Host Address Map

Select **Hosts** from the Left Panel to display the **Manage Host** screen below. Click on **List Hosts.**

Αναγα		Integrated Management SIP Server Management
Help Exit		Server: 192.45.52.160
Top • Users • Conferences	Hanage Ho	sts
Media Server Extensions	List Hosts	List all administered hosts .
Emergency Contacts	Migrate Home/Edge	Migrate a Home/Edge Server.
= Hosts		
List		
Migrate Home/Edge		

Click on the Map link under the 'Commands' heading.

AVAYA				ated Mana Server Man	
Help Exit					02.45.52.160
Top Users Conferences	List Host	ts			
Media Server Extensions	Status (Commands		<u>Host</u>	Туре
Emergency Contacts	up to date Edit Map	Go- Test To Link	Delete	192.45.52.160	home/edge
List					
Migrate Home/Edge Media Servers	Force All Migrate Home/Edg	ge			

Click on the 'Add Map In New Group" item as displayed below.

Αναγα				ed Manager rver Manage	
Help Exit				Server: 192.45	
Top Users Conferences Media Server Extensions		ost Addro .92.45.52.16	-		
Emergency Contacts Hosts	<u>Commands</u>	Name	<u>Commands</u>	<u>Contact</u>	
List Migrate Home/Edge			Edit Delete	sip:\$(user) @192.45.53.100	
• Media Servers	Add Another M	ар	Add Another C	ontact	Delete Group
Adjunct Systems					
Services Server Configuration	Add Map In Ne	w Group			

Add the Host Address Map entry as displayed below. Actual Name and Pattern may vary. In this case, this pattern ('110.*' or 11000 through 11099) is the number sent back to FX-IP over the SIP trunk.

AVAYA		Integrated Management SIP Server Management
Help Exit		Server: 192.45.52.160
Top Users Conferences Media Server Extensions Emergency Contacts Hosts Update All List Migrate Home/Edge Media Servers List	Host 192.45.52.160 Name* FXIPOutbound Pattern* ^sip:110.* Replace URI ✓ Fields marked * are required.	

4.6. Administer Trusted Host

Administer the FX-IP as a trusted host, so that the SIP Bye messages from FX-IP will not be challenged by Avaya SES. To configure a trusted host, use the 'trustedhost -a X - n Y' command in the Linux shell of SES, where 'X' is the IP address of the FX-IP, and 'Y' is the host name of the Avaya SES home server from **Section 4.1**.

```
craft@CCS-DevConl> trustedhost -a 192.45.53.100 -n 192.45.52.160 192.45.53.100 is added to trusted host list.
```

After configuring the trusted host, the user must go back to the Avaya SES administration web interface, and click on the '**Update**' link in the bottom Left Panel for any changes in **Section 4** to take effect.

5. Configure FX-IP

The procedures for configuring the originating and terminating SIP interfaces on FX-IP to interoperate with Avaya Communication Manager will be covered in this section. Use the following steps to configure the FX-IP.

Before continuing with the 'Hammer Configurator', verify that the proper Hammer Services are started. The 'HTServer' and 'DataMgr' services must be operational as below.

<u>File Action View</u>	Help						
💽 🚯	₿ 😫 🕨 🖉 🖬 🕬 👘						
) Services (Local)	🐁 Services (Local)						
	Select an item to view its description.	Name	Description	Status	Startup Type	Log O A	
		🆏 DataMgr		Started	Manual	.\Hammer	
		FxIpServer		Started	Manual	.\Hammer	
		HTIPSigServer			Manual	.\Hammer	
		HTServer 🖏		Started	Manual	.\Hammer	
		Sector LoadSvc		Started	Automatic	.\Hammer	
		🎭 Alerter	Notifies sel		Disabled	Local Service	
		🍓 Application Layer G	Provides s	Started	Manual	Local Service	
		🍓 Remote Registry	Enables re	Started	Automatic	Local Service	
		🍓 Smart Card	Manages a		Manual	Local Service	
		SSDP Discovery Ser	Enables dis	Started	Manual	Local Service	
		🖏 TCP/IP NetBIOS Hel	Enables su	Started	Automatic	Local Service	
		🦓 Uninterruptible Pow	Manages a		Manual	Local Service	
		🤹 Universal Plug and	Provides s		Manual	Local Service	
		🆏 WebClient	Enables Wi	Started	Automatic	Local Service	
		🍓 .NET Runtime Optim	Microsoft		Manual	Local System	
	Extended / Standard /	68. A	manufal a s		Ma1	Caral Contains	

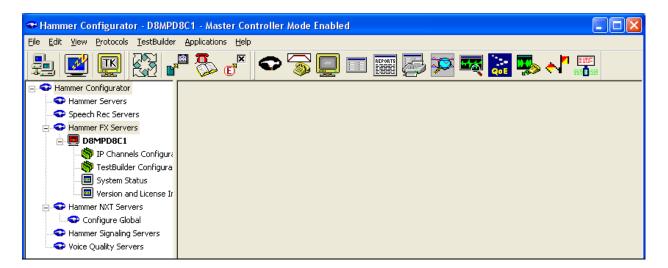
- 1. Start the 'Hammer Configurator' from the 'Start Menu' by selecting Programs → Hammer → Hammer Configurator
- 2. The 'Hammer FX Server' must be configured by highlighting 'Hammer FX Server' on the Right Panel, then click on 'Edit'. Under the 'Edit' pull-down menu, select 'Add/Remover Servers'.



3. 'Enter Server Name'. The server name is usually the Serial Number of the FX-IP. In this case it's 'D8MPD8C1'. Names will vary.

5 Server Licenses	
No Server Licenses	In Use
Enter Server Name:	
	Searching Network
Current Servers:	
Add	
Persona I	
<u>R</u> emove	

4. As a result of entering the Server name, the 'Hammer FX Server' entries should be added as below.



5. On the Left Panel, click on **'IP Channels Configuration**'. The Right Panel opens up displaying the 'Media' tab. Use the 'Media' tab to configure the parameters for sending and receiving media on the channels. To add/change any parameter on each row (or multiple rows by holding down the "Shift' key while selecting), 'right click' on the parameter to be changed.

👁 Hammer Configurator - D8MPD8	BC1 -	Master Contro	oller Mode Enabled					
<u>File E</u> dit <u>V</u> iew <u>P</u> rotocols <u>T</u> estBuilder	Appli	cations <u>H</u> elp						
1 🖉 🖾 🗗	1	E E	> 🛜 📃 i	T 📰 🌄 🦻	ञ 🎫 🔐	1 🐝		
🖃 🗢 Hammer Configurator								
	М	edia Signaling	Options					
Speech Rec Servers			options					
Hammer FX Servers D8MPD8C1		Span : Channels	Audio Xmit Codec	Video Xmit Codec	Media IP Address	Audio Port	T38 Fax Port	Video Port
		0:1-24	G.711 U-Law: 20 ms	Automatic				
👋 TestBuilder Configura		1: 25 - 48 2: 49 - 72	G.711 U-Law: 20 ms G.711 U-Law: 20 ms	Automatic Automatic				
🔤 System Status		3: 73 - 96	G.711 U-Law: 20 ms	Automatic				
Version and License Ir		4: 97 - 120	G.711 U-Law: 20 ms	Automatic				
Hammer NXT Servers		5: 121 - 144	G.711 U-Law: 20 ms	Automatic				
Configure Global		6: 145 - 168	G.711 U-Law: 20 ms	Automatic				
		7: 169 - 192	G.711 U-Law: 20 ms	Automatic				
Hammer Signaling Servers		8: 193 - 216	G.711 U-Law: 20 ms	Automatic				
Servers Voice Quality Servers		9: 217 - 240 10: 241 - 264	G.711 U-Law: 20 ms G.711 U-Law: 20 ms	Automatic Automatic				
		10: 241 - 264 11: 265 - 288	G.711 U-Law: 20 ms G.711 U-Law: 20 ms	Automatic				

When a row on any of the 17 configurable columns is 'right clicked', the configurable information to be entered should be obvious (e.g. Codec, DTMF, enabled / disabled fields) in some cases. Where configurable information is not so obvious, there are two different dialog boxes displayed when 'right clicked'. Select the option "Assign or Increment" to assign a set of values in a particular pattern or incrementer. This Simple Incrementer will be displayed. The following are snapshots of the informational screens provided.

	Simple Incrementer		?
	Server: REM-L-MCHAN	DRA 💌	
	Column:	Phone Name	
	Destination Server.	_	
	Fill Type		
	Channel Fill	ີ Span Fill	
	Tel.Numbers IP Add	dresses URLs MAC Addresses	2
	C IP Address	29.0.0.1 becomes 129.0.1.1)	
		29.0.0.1 becomes 129.0.0.2)	
	C IPv6 (e.g. fe8	0::1[/n][%s] becomes fe80::2[/n][%s]	
	 Trailing Numbers (e.g. User1 becomes User2)	
Select an action:			
Doloce an decioni	Use H323 formattir		
Assign or Increment	Start Value: 110	001	
16 20 20 Table 20 20 20 20 20 20 20 20 20 20 20 20 20	Increment By: 1		
Advanced Increment	Start Channel: 1		
attended for the second	End Channel: 288	8	
Set Field			
	ОКСС	ancel Apply	Help

Set Action Selector

Simple Incrementer

? 🗙

A brief description of each sending / receiving field along with the parameter used during the test
is displayed in the following table:

Option	Description	Configured During Test
Span: Channels	Displays identifiers for the IP spans and channels in the system. You cannot modify this field.	288 Channels were configured (groups of 10)
Audio Xmit Codec	Specifies the transmit codec for audio and the interval between packets in milliseconds.	Automatic, G.711mu, G.729AB, G.723-6.3k
Video Xmit Codec	Specifies the transmit codec for video. This column is only visible if you are licensed for video.	Automatic. Not used
Media IP Address	Specifies the IP address on which each Hammer FX-IP channel will receive the media on, and from which channel it will send media.	192.45.53.100 (IP Address of FX-IP. Will generate and terminate calls over SIP Trunk
Audio Port	Specifies the RTP UDP audio port number in-use at each Media IP Address. This is the port number a channel sends audio from and receives audio on.	Start at 10000 incremented by 2 for each channel.
T.38 Fax	Specifies the UDP port for T.38 fax	Start at 20000

C	SIP and MGCP protocols only). This option is only visible if you are icensed for T.38 Fax.	incremented by 2 for each channel. Not used.
ti n a c	Specifies the UDP port number for he video media. This is the port number a channel sends video from and receives video on. This option is only visible if you are licensed for video.	Start at 22000 incremented by 2 for each channel. Not used.
	Specifies the DTMF transmission nethod.	Both Directions (Inband)
	Specifies the action of an audio tream when it is idle.	GAP
Silence v	Specifies the action of a video stream when it is idle. This column is only visible if you are licensed for video.	GAP
	Allows you to enable or disable a itter buffer.	Disabled
t	Allows you to enable or disable ransmission of RTCP packets with nedia.	Enabled
Profile 7	Specifies the current media profile. The media profile should match the media that you are configuring on the Media Tab.	The name of the .SDP profile under test.
Media IP s p v	Specifies the IP address the media is ent to when using the Clear Channel protocol. This option is only visible when one or more spans are configured for Clear Channel.	Not Used
Audio Port ti C ii s	Specifies the RTP/UDP port to which he media is sent when using the Clear Channel protocol. This option s only visible when one or more pans are configured for Clear Channel.	Not Used
T.38 Port f	Specifies the UDP port to which the ax is sent when using the Clear Channel protocol. This option is only	Not Used
v F	Fax and one or more spans are configured for Clear Channel.	

|--|

6. The codec changes were applied for SIP Trunk testing via the *.SDP file as indicated. The SDP file contains the configuration information on how the FX-IP is setup.

Codec	Media Profile Files
G711	Avaya_Interop_SIP_711.sdp
G723 6.3k	Avaya_Interop_SIP_723.sdp
G729AB	Avaya_Interop_SIP_729.sdp

• The 'Media' tab below was set up for the SIP trunk test as indicated in the snapshot below.

u 🕑 🖳 🐼 🗗	B C	0 2		EFFORTS C	5 🏁		-							
Hammer Configurator														
Servers	Media Signaling	Options												
Speech Rec Servers		Audio Xmit	Video Xmit	Media	Audio	T38 Fax	Video	1	Audio	Video	Jitter		Media	
Hammer FX Servers	Span : Channels	Codec	Codec	IP Address	Port	Port	Port	DTMF	Silence	Silence	Buffer	RTCP	Profile	Call Flow
E REM-L-MCHANDRA	0:1.10	Automatic		192.45.53.100	10000	20000	22000	Both (In-band	Gap		Disabled	Enabled		
IP Channels Configura	1: 11 - 20			192.45.53.100	10000	20000	22000	Both (In-band	Gap	Gap Gap	Disabled	Enabled	2May07_AvayaInteropSIP.sdp 2May07_AvayaInteropSIP.sdp	
👘 TestBuilder Configura	2: 21 - 30			192.45.53.100	10020	20020	22020	Both (In-band	Gap	Gap	Disabled	Enabled	2May07_AvayaInteropSIP.sdp	Audio Onl
System Status	3: 31 - 40			192.45.53.100	10040	20040	22040	Both (In-band	Gap	Gap		Enabled	2May07_AvayaInteropSIP.sdp	
Version and License Ir	4: 41 - 50	Automatic		192.45.53.100	10080	20080	22080	Both (In-band	Gap	Gap	Disabled	Enabled	2May07_AvayaInteropSIP.sdp	Audio Onl
	5: 51 - 60	Automatic		192,45,53,100	10100	20100	22100	Both (In-band	Gap	Gap	Disabled	Enabled	2May07 AvayaInteropSIP.sdp	Audio Onl
Ammer NXT Servers	6: 61 - 70	Automatic	Automatic	192.45.53.100	10120	20120	22120	Both (In-band	Gap	Gap	Disabled	Enabled	2May07_AvayaInteropSIP.sdp	Audio Onl
- Configure Global	7: 71 - 80	Automatic	Automatic	192.45.53.100	10140	20140	22140	Both (In-band	Gap	Gap	Disabled	Enabled	2May07 AvayaInteropSIP.sdp	Audio Onl
Hammer Signaling Servers	8: 81 - 90	Automatic	Automatic	192.45.53.100	10160	20160	22160	Both (In-band	Gap	Gap	Disabled	Enabled	2May07_AvayaInteropSIP.sdp	Audio Onl
Servers	9:91 - 100	Automatic			10180	20180	22180	Both (In-band	Gap	Gap	Disabled	Enabled	2May07_AvayaInteropSIP.sdp	Audio On
 Voice Quality Servers 	10: 101 - 110			192.45.53.100	10200	20200	22200	Both (In-band	Gap	Gap	Disabled	Enabled		Audio Onl
	11:111 - 120	Automatic		192.45.53.100	10220	20220	22220	Both (In-band	Gap	Gap	Disabled	Enabled	2May07_AvayaInteropSIP.sdp	
	12: 121 - 130	Automatic		192.45.53.100	10240	20240	22240	Both (In-band	Gap	Gap	Disabled	Enabled	2May07_AvayaInteropSIP.sdp	Audio Onl
	13: 131 - 140	Automatic		192.45.53.100	10260	20260	22260	Both (In-band	Gap	Gap	Disabled	Enabled		Audio Onl
	14: 141 - 150			192.45.53.100	10280	20280	22280	Both (In-band	Gap	Gap	Disabled	Enabled	2May07_AvayaInteropSIP.sdp	
	15: 151 - 160			192.45.53.100	10300	20300	22300	Both (In-band	Gap	Gap	Disabled	Enabled	2May07_AvayaInteropSIP.sdp	
	16: 161 - 170 17: 171 - 180	Automatic Automatic		192.45.53.100 192.45.53.100	10320	20320 20340	22320 22340	Both (In-band Both (In-band	Gap Gap	Gap	Disabled Disabled	Enabled	2May07_AvayaInteropSIP.sdp 2May07_AvayaInteropSIP.sdp	
	18: 181 - 190			192.45.53.100	10340	20340	22340	Both (In-band	Gap	Gap Gap	Disabled	Enabled	2May07_AvayaInteropSIP.sdp 2May07_AvayaInteropSIP.sdp	
	19: 191 - 200	Automatic	Automatic	192.45.53.100	10360	20360	22360	Both (In-band	Gap	Gap	Disabled	Enabled	2May07_AvayaInteropSIP.sdp 2May07 AvayaInteropSIP.sdp	Audio Oni
	20: 201 - 210	Automatic	Automatic	192.45.53.100	10300	20300	22400	Both (In-band	Gap	Gap	Disabled	Enabled	2May07_AvayaInteropSIP.sdp 2May07_AvayaInteropSIP.sdp	Audio Onl
	20.201-210	Automatic	Automatic		10400	20400	22400	Ded. (I., Land	C	C	Disableu	Faction J		Audio Only
	Configuration Manag		7_CM_SIP_2	0ch_G711_20	•									

7. Configure the protocol and signaling on each channel by clicking the 'Signaling' tab. To select multiple rows do a Shift select by holding down the "Shift' key while selecting.

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Hammer Configurator - D8MPD8C1 - File Edit View Protocols TestBuilder Applic		de Enabled		
ᆋ 🛃 🔛 🐼 🗗 🖏		5 📃 💷 📱	🎬 🛃 🌠 🐺	: 🍢
Hammer Configurator Hammer Servers Hammer Servers Hammer FX Serve	edia Signaling Options Basic SIP MGCP	H323 Skinny		
 IP Channels Configure TestBuilder Configura System Status Version and License Ir Hammer NXT Servers Configure Global Hammer Signaling Servers Voice Quality Servers 	Channels Channel: 1 Channel: 2 Channel: 3 Channel: 4 Channel: 5 Channel: 6 Channel: 6 Channel: 7 Channel: 8 Channel: 9 Channel: 9 Channel: 10 Channel: 11 Channel: 12 Channel: 14	Protocol SIP	State Machi Avaya SIP Phone Avaya SIP Phone	ne
	ifiguration Management Load: Itrain Apply New	ing Save Help Help		
unches the Signaling Editor application				50

• Explanation of the Columns under the Signaling \rightarrow Basic

Parameters	Descriptions	Test Value
Channels	Displays identifiers for the IP channels in the	288 channels. Only
	system, one channel per row. You cannot modify	20 used.
	this field.	
Protocol	Select the protocol that provides IP signaling. You	SIP
	must assign the protocol within span boundaries;	
	you cannot assign a protocol on individual	
	channels. To select the protocol, right-click and	
	select one of the following protocols from the	
	drop-down list:	
State Machine	Select the state machine that defines the behavior	Empirix Modified to
	of an emulated Trunk. Right-click and select a	accommodate
	name from the drop-down menu, or select Set	'shuffling'.
	Field and specify a state machine name in the text	Otherwise, select

.box.	'SIP Phone'

- 8. The 'Signaling \rightarrow SIP' tab is used to configure the SIP parameters. The following two screen shots display the parameters configured for the SIP Trunk test.
- Screen Shot 1

Channels	Phone Name	Display Name	Phone IP	Phone Port	Destination Address (e.g. Proxy)	Destination Port	Outbound Proxy Address	Outbound Proxy Port	Authentication Name	Authentication Password	Register with Proxy?	Requested Expiration [s]	Auto re- Register
Channel: 1	11001	Channel1	192.45.53.100	5060	192.45.52.160	5060		1	11001	123456	No	3600	No
Channel: 2	11002	Channel2	192,45,53,100	5060	192,45,52,160	5060			11002	123456	No	3600	No
Channel: 3	11003	Channel3	192.45.53.100	5060	192.45.52.160	5060			11003	123456	No	3600	No
Channel: 4	11004	Channel4	192.45.53.100	5060	192.45.52.160	5060			11004	123456	No	3600	No
Channel: 5	11005	Channel5	192.45.53.100	5060	192.45.52.160	5060			11005	123456	No	3600	No
Channel: 6	11006	Channel6	192.45.53.100	5060	192.45.52.160	5060			11006	123456	No	3600	No
Channel: 7	11007	Channel7	192.45.53.100	5060	192.45.52.160	5060			11007	123456	No	3600	No
Channel: 8	11008	Channel8	192.45.53.100	5060	192.45.52.160	5060			11008	123456	No	3600	No
Channel: 9	11009	Channel9	192.45.53.100	5060	192.45.52.160	5060			11009	123456	No	3600	No
Channel: 10	11010	Channel10	192.45.53.100	5060	192.45.52.160	5060			11010	123456	No	3600	No
Channel: 11	11011	Channel11	192.45.53.100	5060	192.45.52.160	5060			11011	123456	No	3600	No
Channel: 12	11012	Channel12	192.45.53.100	5060	192.45.52.160	5060			11012	123456	No	3600	No
Channel: 13	11013	Channel13	192.45.53.100	5060	192.45.52.160	5060			11013	123456	No	3600	No
Channel: 14	11014	Channel14	192.45.53.100	5060	192.45.52.160	5060			11014	123456	No	3600	No
Channel: 15	11015	Channel15	192.45.53.100	5060	192.45.52.160	5060			11015	123456	No	3600	No
Channel: 16	11016	Channel16	192.45.53.100	5060	192.45.52.160	5060			11016	123456	No	3600	No
Channel: 17	11017	Channel17	192.45.53.100	5060	192.45.52.160	5060			11017	123456	No	3600	No
Channel: 18	11018	Channel18	192.45.53.100	5060	192.45.52.160	5060			11018	123456	No	3600	No
Channel: 19	11019	Channel19	192.45.53.100	5060	192.45.52.160	5060			11019	123456	No	3600	No
Channel: 20	11020	Channel20	192.45.53.100	5060	192.45.52.160	5060			11020	123456	No	3600	No
Channel: 21	11021	Channel21	192.45.53.100	5060	192.45.52.160	5060			11021	123456	No	3600	No
Channel: 22	11022	Channel22	192.45.53.100	5060	192.45.52.160	5060			11022	123456	No	3600	No
Channel: 23	11023	Channel23	192.45.53.100	5060	192.45.52.160	5060			11023	123456	No	3600	No
Channel: 24	11024	Channel24	192.45.53.100	5060	192.45.52.160	5060			11024	123456	No	3600	No
Channel: 25	11025	Channel25	192.45.53.100	5060	192.45.52.160	5060			11025	123456	No	3600	No
Channel: 26	11026	Channel26	192.45.53.100	5060	192.45.52.160	5060			11026	123456	No	3600	No
Channel: 27	11027	Channel27	192.45.53.100	5060	192.45.52.160	5060			11027	123456	No	3600	No

• Screen Shot 2 (this is a continuation of Screen Shot 1)

Registration Stagger [ms]	Require PRACK	Transport Protocol	Support Session Timer	Request Session Timer	Refresher	Session Expires	Min SE	Use Update	Use Shuffling
100	No	UDP	No	No	none	1800	90	No	No
200	No	UDP	No	No	none	1800	90	No	No
300	No	UDP	No	No	none	1800	90	No	No
400	No	UDP	No	No	none	1800	90	No	No
500	No	UDP	No	No	none	1800	90	No	No
600	No	UDP	No	No	none	1800	90	No	No
700	No	UDP	No	No	none	1800	90	No	No
800	No	UDP	No	No	none	1800	90	No	No
900	No	UDP	No	No	none	1800	90	No	No
1000	No	UDP	No	No	none	1800	90	No	No
1100	No	UDP	No	No	none	1800	90	No	No
1200	No	UDP	No	No	none	1800	90	No	No
1300	No	UDP	No	No	none	1800	90	No	No
1400	No	UDP	No	No	none	1800	90	No	No
1500	No	UDP	No	No	none	1800	90	No	No
1600	No	UDP	No	No	none	1800	90	No	No
1700	No	UDP	No	No	none	1800	90	No	No
1800	No	UDP	No	No	none	1800	90	No	No
1900	No	UDP	No	No	none	1800	90	No	No
2000	No	UDP	No	No	none	1800	90	No	No
2100	No	UDP	No	No	none	1800	90	No	No
2200	No	UDP	No	No	none	1800	90	No	No
2300	No	UDP	No	No	none	1800	90	No	No
2400	No	UDP	No	No	none	1800	90	No	No
2500	No	UDP	No	No	none	1800	90	No	No
2600	No	UDP	No	No	none	1800	90	No	No
2700	No	UDP	No	No	none	1800	90	No	No
2700	NO	UUP	No	NO	none	1800	90	NO	No

Signaling Configuration for Trunk

Note: Last Column was created by Empirix to allow for 'Shuffling' to be toggled on or off.

• SIP Tab explanation.

This section is used to configure the endpoint or the trunk with following parameters.

Parameters	Required	Description
Channels	N/A	Displays identifiers for the channels. This
		field cannot be modified.
Phone Name	Yes	Defines the user name of the endpoint. For
		example, 56145 or phone01. You can use the
		Simple Incrementer or the Advanced
		Incrementer to enter a range of values.
Display Name	No	Defines the name to be displayed in the user
		interface of the far endpoint.
Phone IP	Yes	The IP address used for sending and
		listening. This must be a valid IPv4 address
		or a fully qualified domain name. This field
		must resolve to the same address as the
DI D (X 7	Local Signaling IP field on the Basic tab.
Phone Port	Yes	The SIP UDP port in use at each Phone IP
		address for sending and listening for SIP
Destination	V	messages. IP address or domain name of each channel's
Destination Address	Yes	
Audress		SIP proxy server or, if not using a proxy server, the IP address of a channel's
		destination. This address is used in the
		request URI (the destination address in the
		request). While using a SIP proxy server is
		not required, a value is required for this
		field.
Destination	Yes	The SIP UDP port in-use at each Proxy IP
Port	100	address for sending and listening for SIP
		messages.
Authentication	No	Name used in the username/password
Name		authentication scheme.
Authentication	No	Password used in the username/password
Password		authentication scheme.
Register with	No	Select whether or not a channel
Proxy?		automatically registers with a SIP proxy
		server when the channel initializes. If
		registering with a proxy, enter the proxy's
		address in the Destination Address field and
		its SIP port in the Destination Port field.

Requested Expiration	Required if registering with a proxy.	Requested time in seconds that the proxy should keep a channel registration active. The requested expiration is sent in the channel's REGISTER messages. The receiving endpoint can accept the requested value or respond with a lower value.
Autos re- Register?	Required if registering with a proxy.	Select whether to automatically reregister channels with the SIP proxy. Right-click and select one of the options.
Use Shuffling	No	Select whether to use shuffling or no shuffling of media. Note: Empirix Modified

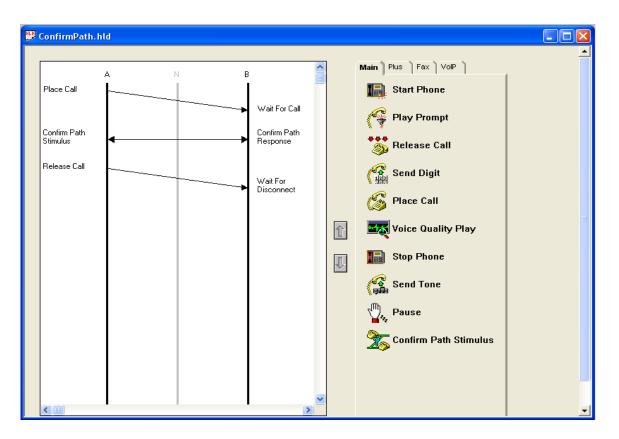
6. Empirix Test Builder

TestBuilder is a telephony testing software package that allows you to easily create and run load tests using a simple graphical interface.

TestBuilder provides the following:

- Two interfaces for creating tests. You can assemble test action icons in a TestBuilder ladder diagram, or write a test script using Hammer Visual Basic (HVB) in the HVB editor.
- Test monitors show test statistics and channel status in real-time as you run a test.
- Reports created after a test finishes provide detail for an entire test and for individual channels.
- Scheduling options including the ability to run multiple tests simultaneously
- TestBuilder Plus includes predefined calling patterns that enable you to simulate real world operating conditions. You can schedule a test on a group of channels and allow the Hammer to control when each channel starts in order to create the calling pattern you selected.

Use a predefined test case with bidirectional voice path confirmation from A to B and vice versa. This test case is executed throughout the interoperability testing with Avaya Communication Manager.



A 'Right Click' with the mouse on any of the above actions brings up an additional dialog box. For example, a 'Right Click' on 'Place Call' or 'Confirm Path Stimulus' respectively, displays the following:

Place Call Properties	
Phone Number © Use <u>D</u> ial String	OK Cancel Help
 ○ Use <u>Phone book</u> ○ Use <u>Channel Map</u> 	
Protocol Parameters	,
ISDN SS7	H.323
Connect Latency Do Connect Latency Connect Latency	tency Params

Place Call Properties settings

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Confirm Path	X
Media Tones A-Side: F1: 700 F2: 800 F3: 900 On-Time (ms): 500 B-Side: F1: 700 F2: 800 F3: 900 On-Time (ms): 500 Image: Solid constraints Image: Solid constraints Image: Solid constraints Media Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints Image: Solid constraints <td>OK Cancel Help</td>	OK Cancel Help
B-Side: Prompt Name: Hello.711u Encoding: G711U Call Terminator Digit Digit Terminator: Timeout Override Timeout (ms): 10000	
Call Hold Time Random Minimum Hold Time (s): Image: State S	

Confirm Path using Voice Prompts for G711U

7. Phone Book Setup

Use the PhoneBook Settings tab to enter phone numbers to call. You can enter numbers on a channel or span basis. If you enter one number for an entire span, all channels on that span dial that number. **Note:** A 'Span' is a configurable number of channels. To create a Phone Book apply the following steps:

1. Open the Hammer Configurator and verify the Master Controller mode is enabled as indicated on the top line of 'Hammer Configurator' application window.

👁 Hammer Configurator - D8MPI	D8C1 - Master Controller Mode Enabled	
<u>File E</u> dit <u>V</u> iew <u>P</u> rotocols <u>T</u> estBuilder	Applications Help	
	" 🐉 🗗 🗢 🕱 🛄 🗉 🧱 👺 🐺 😽 📲	
Hammer Configurator	PhoneBook	^
Hammer Servers		
Speech Rec Servers		
🚊 🖘 Hammer FX Servers		=
🖶 🖘 Hammer NXT Servers		
🗢 😔 Configure Global		_
Servers Voice Quality Servers		
		~
Displays the TestBuilder PhoneBook		SCRL /

2. Open the PhoneBook. by selecting TestBuilder → PhoneBook on the Hammer Configurator Menu. The last saved PhoneBook opens. If the last saved PhoneBook cannot be found, a new blank PhoneBook opens.

→ 🖹 → 🏢 🛛 🖬	1 1 1			
honebook Settings Use Phone List	Channel Map Settings Phone List Settings	Phone #	Field 1	Field 2
	REM-L-MCHANDRA Span0:1-10			
_				
	REM-L-MCHANDRA Span1:11-20			
	REM-L-MCHANDRA Span1:11-20 REM-L-MCHANDRA Span2:21-30			
	REM-L-MCHANDRA Span2:21-30			
	REM-L-MCHANDRA Span2:21-30 REM-L-MCHANDRA Span3:31-40			
	REM-L-MCHANDRA Span2:21-30 REM-L-MCHANDRA Span3:31-40 REM-L-MCHANDRA Span4:41-50			
	REM-L-MCHANDRA Span2:21-30 REM-L-MCHANDRA Span3:31-40 REM-L-MCHANDRA Span4:41-50 REM-L-MCHANDRA Span5:51-60			

To open a different saved PhoneBook, select File, Open, and then select a PhoneBook (.phn) file. To open a new PhoneBook, select File, New.

- 3. Select the Phonebook Settings tab.
- 4. Using one or more of the following methods, enter dialing information for each channel that will place a call in the SIP Trunk test. Enter phone numbers in the Phone # column. Optionally, enter multi-stage dialing digits in the Field 1, Field 2, and Field 3 columns.
- Phone book used for SIP Trunk Testing.

→	≆ 🖬 🔸			
onebook Settings	Channel Map Settings Phone List Settings			
Use Phone List	Channel	Phone #	Field 1	Field 2 🔺
	REM-L-MCHANDRA Span0:1	811011		
	REM-L-MCHANDRA Span0:2	811012		
	REM-L-MCHANDRA Span0:3	811013		
	REM-L-MCHANDRA Span0:4	811014		
	REM-L-MCHANDRA Span0:5	811015		
	REM-L-MCHANDRA Span0:6	811016		
	REM-L-MCHANDRA Span0:7	811017		
	REM-L-MCHANDRA Span0:8	811018		
	REM-L-MCHANDRA Span0:9	811019		
	REM-L-MCHANDRA Span0:10	811020		
	REM-L-MCHANDRA Span1:11-20			
	REM-L-MCHANDRA Span2:21-30			
	REM-L-MCHANDRA Span3:31-40			
	REM-L-MCHANDRA Span4:41-50			
	REM-L-MCHANDRA Span5:51-60			
	REM-L-MCHANDRA Span6:61-70			
	REM-L-MCHANDRA Span7:71-80			
	REM-L-MCHANDRA Span8:81-90			
	REM-L-MCHANDRA Span9:91-100			
	REM-L-MCHANDRA Span10:101-110			-
•				•

Option	Method
To create a single entry for an entire span:	Enable the Propagate function and enter digits in the columns next to the span name.
To create individual entries for each channel one by one:	Disable the Propagate function, click a span to expand its list of channels, and enter digits in the columns next to each channel.
To enter a range of values for a selected group of channels or spans:	 Right-click a channel and select one of the following options: Increment using a simple format. Increment using an advanced formula.

- 5. When finished entering values, select File, Save. The Save PhoneBook dialog box appears.
- 6. Enter a file name and click Save. If prompted to overwrite an existing PhoneBook, click Yes. PhoneBook files (including .phn, .map, and .lst) are saved in the \Hammer\LoadBlaster\Config\GlobalPhoneBooks directory.
- 7. When asked if you want to replace the TestBuilder default PhoneBook with the latest saved PhoneBook, click Yes. Otherwise, click No.

If you click Yes, the system copies the PhoneBook files to every Hammer server connected to your system. These files become the default PhoneBook. If tests are running, the new default PhoneBook files will take effect after all tests have stopped.

If you click No, you can replace TestBuilder with the latest saved PhoneBook files later by opening the saved PhoneBook and selecting Update on the Hammer Configurator TestBuilder menu.

8. Select File, Exit to close the PhoneBook Suite window.

8. Interoperability Compliance Testing

The Interoperability compliance testing focused on the following areas in FX-IP:

- Generation of moderate SIP telephony load to Avaya Communication Manager via the trunk interface to Avaya SES, and back out to the SIP users on FX-IP from Avaya SES.
- Support of various SIP audio codecs with Avaya Communication Manager and Avaya SES.
- Support of non-direct audio, and direct audio with media shuffling.
- Recovery from adverse conditions during the load test.

8.1. General Test Approach

The feature test cases were conducted by using FX-IP to originate and terminate SIP calls to Avaya Communication Manager via Avaya SES. The audio codec test calls were held up for 65 seconds. The serviceability test cases were performed by disconnecting and reconnecting the LAN cables on the FX-IP.

The verification included monitoring the various reports from FX-IP during and after the test runs, and checking the status of various SIP resources on Avaya Communication Manager and Avaya SES.

8.2. Test Results

All test cases were executed and passed. SIP Trunking without Audio Shuffling functioned with the FX-IP 'SIP Default' settings.

Additional FX-IP customization was required to support IP Direct Media (aka – Audio Shuffling) and can be available through Empirix.

9. Verification Steps

This section provides the tests that can be performed to verify proper configuration of SIP between Avaya Communication Manager and FX-IP.

9.1. Verify Avaya Communication Manager

Verify the status of the SIP trunk group by using the 'status trunk n' command, where 'n' is the trunk group number administered in Section 3.5. Verify all trunks are in the 'in-service/active' state as shown below.

```
status trunk 10
                          TRUNK GROUP STATUS
Member Port Service State
                               Mtce Connected Ports
                                 Busy
0010/001 T00173 in-service/active no
                                     T00177
0010/002 T00174 in-service/idle
                                 no
0010/003 T00175 in-service/idle
                                 no
0010/004 T00176 in-service/idle no
0010/005 T00177 in-service/active no
                                     T00173
0010/006 T00178 in-service/idle no
0010/007 T00179 in-service/idle no
0010/008 T00180 in-service/active no
                                     T00182
0010/009 T00181 in-service/idle no
0010/010 T00182 in-service/active no
                                     T00180
```

Verify the status of the SIP signaling group by using the 'status signaling-group n' command, where 'n' is the signaling group number administered in **Section 3.6**. Verify the signaling group is 'in-service' as indicated in the Group State field shown below.

```
status signaling-group 10
                        STATUS SIGNALING GROUP
       Group ID: 10
                                                  Active NCA-TSC Count: 0
     Group Type: sip
                                                  Active CA-TSC Count: 0
 Signaling Type: facility associated signaling
    Group State: in-service
```

Verify the status of the SIP station by using the 'status station n' command, where 'n' is the extension of an administered SIP station from Section 3.8. Verify the Off-PBX Service State is 'in-service/active'.

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```
status station 11001
                                                                     1 of
                                                                            3
                                                              Page
                             GENERAL STATUS
    Administered Type: 6408D+ Service State: No hardware assigned
       Connected Type: N/A
                                   Parameter Download: pending
            Extension: 11001
                                   SAC Activated? no
User Cntrl Restr: none
                 Port: X
          Call Parked? no
                                     Group Cntrl Restr: none
     Ring Cut Off Act? no
                                    CF Destination Ext:
Active Coverage Option: 1
                                Off-PBX Service State: in-service/active
         EC500 Status: N/A
      Message Waiting:
  Connected Ports:
                                             HOSPITALITY STATUS
                                           Awaken at:
                                           User DND: not activated
                                           Group DND: not activated
                                         Room Status: non-guest room
```

Verify the status of a connected SIP trunk by using the 'status trunk x/y', where 'x' is the number of the SIP trunk group from **Section 3.5** and 'y' is the member number of a connected trunk. Verify the **Service State** is 'in-service/active', and that the IP addresses of the C-LAN and SES server are shown in the **Signaling** section. In addition, the **Audio** section shows the codec type and the IP address of the FX-IP. The **Audio Connection Type** displays 'ip-direct', indicating media shuffling.

```
status trunk 10/1
                                                              Page
                                                                     1 of
                                                                            2
                             TRUNK STATUS
Trunk Group/Member: 0010/001
                                         Service State: in-service/active
                                    Maintenance Busy? no
              Port: T00173
Signaling Group ID:
   Connected Ports: T00177
                  Port
                         Near-end IP Addr : Port
                                                   Far-end IP Addr : Port
       Signaling: 01A0617 192. 45.100.147 : 5061 192. 45. 53.100 : 5061
G.711MU
           Audio:
                          192. 45. 53.100 : 6000
                                                     192. 45. 53.100 : 6000
           Video:
     Video Codec:
                                          Authentication Type: None
   Audio Connection Type: ip-direct
```

9.2. Verify Avaya SIP Enablement Services

From the Linux shell of SES, use the 'trustedhost –L' command to verify the IP address of the FX-IP is listed as a trusted host.

craft@CCS-DevCon1> trustedh Third party trusted hosts.	nost -L	
Trusted Host	CCS Host Name	Comment
192.45.53.100	192.45.52.160	

9.3. Empirix FX-IP Test Execution

Using the 'Hammer TestBuilder', right click on the 'test case name' from the Right Panel. The 'test case name' will vary. Click on 'Schedule' followed by 'Edit & Run'.



The Scheduler Window will popup and the screen shown below is displayed. The SIP Trunk tests were setup with the information displayed. At this point, clicking 'OK' starts the test.

Properties	
TB Scheduler Other	
er\Library\Hammer\CallProfileTests\ConfirmPath.hld Start Time: 11:11:42 AM 5/ 8/2007	Action if a Channel is busy:
Channels A-Side: REM-L-MCHANDRA[1-10] B-Side: REM-L-MCHANDRA[11-20]	Max <u>Active Connections:</u> (0 = Unlimited)
Stagger	Max Test Time: Hours: 0
User Defined - (ms) 100 Min (s) 1	Loop Count: (-1 = Loop Forever)
C Random - Max (s) 5 C None	<u>G</u> uard Time (ms): 5000
OK	Cancel Apply Help

• Select the channels to use to execute the test for 'A-Side' and 'B-Side' as shown in figure below.

Channel Browser		X
Available Channels:	A-Side:	<u> </u>
REM-L-MCHANDRA Span: 2 Span: 3 Span: 4 Span: 5 Span: 6 Span: 7 Span: 9 Span: 10 Span: 12 Span: 13 Span: 14 Span: 13 Span: 14 Span: 15 Span: 16 Span: 17 Span: 18 Span: 19	Map 4: SIP Channel	

9.4. Report Generation

To view the reporting processes, from the 'Hammer – TestBuilder', select 'Report' from the Applications pull-down Menu.

To generate a report, you first select the report type from the drop-down list in the top left-hand corner of the Report window. A dialog box appears that allows you to select parameters for the report, including the Test Case Name, time and test data range.

P Metrics		- 8	🗅 🖄 😭	🗟 🗖	🔳 🦉	3 💢 🖥	ह्यू 🛜 🦉	🍌 💎 🖁				
I	• •		100% •	🚧 Total:	41 100%	41 of 41	P.					_
				RTP METRI	CSREPORT							
				May/8/2007	10:00:50	AM				6		
Test Name:	Confirm	nPath_A.sbx										
Start Time:	2007-05	-02 18:48:0	0									
Stop Time:	2007-05	-02.18-52-5	9									
S top Time:		-02 18:52:5 and 59 second										
Stop Time:		-02 18:52:5 and 59 second			<u>Transmit</u>		<u>Receive</u>					
S top Time: <u>TimeStamp</u>				<u>Media</u>	<u>Transmit</u> BytesOut	<u>Pkts Out</u>	<u>Receive</u> BytesIn	<u>Phish</u>	<u>005eq</u>	Lost	<u>Dupe</u>	e
-	4 minutes	and 59 second	ls	<u>Media</u> G <i>3</i> 11u		Pkis0ut 839		<u>Phistra</u> 3,450	<u>005eq</u> 0	<u>Lost</u> 0	<u>Dupe</u> 0	e
<u>TimeStamp</u>	4 minutes <u>Server</u>	and 59 second <u>Chan#</u>	2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2		<u>By tes Out</u>		<u>By tes In</u>					ŧ
<u>TimeStamp</u> 5/2/2007 6:48:14PM	4 minutes Server H4911	and 59 second <u>Chan#</u> 6	is <u>Con 10</u> 000000020100006	6711u	<u>By tes Out</u> 143,372	839	By tes In 593,400	3,450	0	0	0	ŧ
<u>TimeStamp</u> 5/2/2007 6:48:14PM 5/2/2007 6:48:14PM	4 minutes <u>Server</u> H4911 H4911	and 59 second <u>Chan#</u> 6 2	s <u>C vil 10)</u> 000000020100006 00000020100002	6.711u 6.711u	By tes Out 143,372 143,372	839 839	BytesIn 593,400 594,432	3,450 3,456	0	0	0 0	<u>e</u>
TimeStamp 5/2/2007 6:48:14PM 5/2/2007 6:48:14PM 5/2/2007 6:48:15PM	4 minutes <u>Server</u> H4911 H4911 H4911	and 59 second <u>Chan#</u> 6 2 1	s <u>Can m</u> 000000020100006 000000020100002 000000020100001	6.711u 6.711u 6.711u	<u>BytesOut</u> 143,372 143,372 143,372	839 839 839	Bytesin 593,400 594,432 596,840	3,450 3,456 3,470	0 0 0	0 1 1	0 0 0	<u>e</u>
TimeStamp 5/2/2007 6:48:14PM 5/2/2007 6:48:14PM 5/2/2007 6:48:15PM 5/2/2007 6:48:15PM	4 minutes <u>Server</u> H4911 H4911 H4911 H4911	and 59 second <u>Chan#</u> 6 2 1 9	E SET 10 C SET 10 000000020100006 000000020100002 000000020100001 000000020100009	6711u 6711u 6711u 6711u	By tes Out 143,372 143,372 143,372 143,372	839 839 839 839	By tes In. 593,400 594,432 596,840 593,228	3,450 3,456 3,470 3,449	0 0 0	0 1 1 0	0 0 0	ŧ
TimeStamp 5/2/2007 6:48:14PM 5/2/2007 6:48:14PM 5/2/2007 6:48:15PM 5/2/2007 6:48:15PM 5/2/2007 6:48:15PM	4 minutes <u>Server</u> H4911 H4911 H4911 H4911 H4911	sand 59 second <u>Chan#</u> 6 2 1 9 7	Call ID Color 000000020100006 000000020100002 000000020100001 000000020100009 000000020100007	6711u 6711u 6711u 6711u 6711u	By tes Out 143,372 143,372 143,372 143,372 143,372	839 839 839 839 839 839	By tes In. 593,400 594,432 596,840 593,228 593,200	3,450 3,456 3,470 3,449 3,450	0 0 0 0	0 1 1 0 0	0 0 0 0	<u>e</u>
TimeSfamp 5/2/2007 6:49:14PM 5/2/2007 6:49:14PM 5/2/2007 6:49:15PM 5/2/2007 6:49:15PM 5/2/2007 6:49:16PM 5/2/2007 6:49:16PM	4 minutes Server H4911 H4911 H4911 H4911 H4911 H4911	and 59 second <u>Chan#</u> 6 2 1 9 7 5	s c al 10 00000020100006 00000020100002 00000020100009 00000020100009 000000020100007	6711u 6711u 6711u 6711u 6711u 6711u	Bytes Out 143,372 143,372 143,372 143,372 143,372 143,372 143,372	839 839 839 839 839 839 839	By tes In. 593,400 594,432 596,840 593,228 593,200 593,744	3,450 3,456 3,470 3,449 3,450 3,452	0 0 0 0 0	0 1 1 0 0	0 0 0 0 0	e
TimeStamp 5/2/2007 6:48:14PM 5/2/2007 6:48:14PM 5/2/2007 6:48:15PM 5/2/2007 6:48:15PM 5/2/2007 6:48:16PM 5/2/2007 6:48:16PM	4 minutes Server H4911 H4911 H4911 H4911 H4911 H4911 H4911	and 59 second <u>Chan#</u> 6 2 1 9 7 5 3	c ul 10	6711u 6711u 6711u 6711u 6711u 6711u 6711u	BytesOut 143,372 143,372 143,372 143,372 143,372 143,372 143,372 143,372	839 839 839 839 839 839 839 839	By tes In. 593,400 594,432 596,840 593,228 593,400 593,744 594,088	3,450 3,456 3,470 3,449 3,450 3,452 3,452	0 0 0 0 0 0	0 1 1 0 0 0	0 0 0 0 0 0	<u>e</u>
TimeStang 5/2/2007 6:48:14PM 5/2/2007 6:48:14PM 5/2/2007 6:48:15PM 5/2/2007 6:48:15PM 5/2/2007 6:48:16PM 5/2/2007 6:48:16PM 5/2/2007 6:48:16PM	4 minutes 5 erver H4911 H4911 H4911 H4911 H4911 H4911 H4911 H4911	and 59 second <u>Chan#</u> 6 2 1 9 7 5 3 8	c al 10 000000020100006 00000020100002 000000020100001 000000020100007 000000020100005 000000020100005 000000020100003	6711u 6711u 6711u 6711u 6711u 6711u 6711u 6711u	BytesOut 143,372 143,372 143,372 143,372 143,372 143,372 143,372 143,372 143,372	839 839 839 839 839 839 839 839 839	By tes In. 593,400 594,432 596,840 593,228 593,400 593,744 594,088 594,432	3,450 3,456 3,470 3,449 3,450 3,452 3,454 3,456	0 0 0 0 0 0 0	0 1 1 0 0 0 0 0	0 0 0 0 0 0 0	<u>e</u>

The 'RTP Metrics Report' above is an example of the results of a test run. The 'Hammer Reports' tool can be fine tuned to key-on various parameters for a particular need.

10. Support

Technical support on Empirix FX Base IP can be obtained through the following:

- Email the Empirix support center via <u>support@empirix.com</u>.
- Call the Empirix support center at 1-800-Empirix.

11. Conclusion

These Application Notes describe the configuration steps required for Empirix FX-IP Release 2.4.1 to successfully interoperate with Avaya Communication Manager 4.0 and Avaya SIP Enablement Services 3.1.2. All feature and serviceability test cases were completed successfully.

12. Additional References

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

[1] Administrator Guide for Avaya Communication Manager, Document 03-300509, Issue 1, June 2005, available at <u>http://support.avaya.com</u>

[2] SIP Enablement Services R3.0 Installation and Administration Guide,, Issue 5.1, July 2005, available at <u>http://support.avaya.com</u>

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