Avaya Solution & Interoperability Test Lab

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

Abstract

These Application Notes describe the configuration steps required for Empirix FX-IP (FX-IP) to interoperate with Avaya Communication Manager using the SIP Endpoint 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.

1. 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, the FX-IP originated and terminated the calls to Avaya SES. To accomplish the SIP endpoint emulation between the two systems, the following was performed:

- Emulated registered FX-IP endpoints, each with its own individual IP address, registered with Avaya SES and generate calls from each endpoint
- Avaya Communication Manager receives the calls and redirects through Automatic Alternate Routing (AAR)
- Terminate calls at the FX-IP individual IP addressed emulated endpoint

1.1. Registered Endpoint Flow

The registered SIP endpoint calls originating from FX-IP travel over the SIP trunks to Avaya SES. The Avaya SES passes the calls via separate SIP trunks to the C-LAN card on the Avaya G650 Media Gateway. Avaya Communication Manager then routes the calls back out to Avaya SES over the SIP trunks, and the Avaya SES passes the calls to the FX-IP where the registered SIP endpoint calls are terminated.

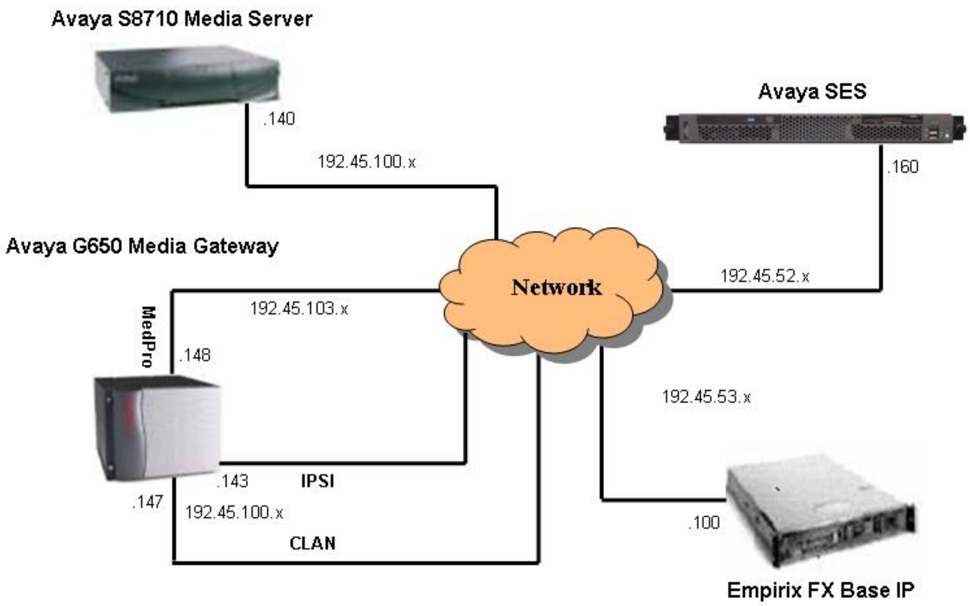


Figure 1: Network Configuration

1.2. FX-IP Capacity

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

The FX-IP capacities are licensing based, however the Empirix FX-Base-IP 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 S8700 Media Servers	Avaya Communication Manager 4.0.0 (R014x.00.0.730.5)
Avaya MCC1 Media Gateway • TN799DP C-LAN Circuit Pack • TN2302AP IP Media Processor Circuit Pack • TN2312BP IP Server Interface	 HW01 FW015 HW13 FW095 HW12 FW039
Avaya C364T-PWR Converged Stackable Switch	4.5.14
Avaya SIP Enablement Services	3.1.2 (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 Licenses
- Administer IP codec set and network region
- Administer IP node names for C-LAN and Avaya SES server
- Administer IP interface and data module for C-LAN
- Administer SIP trunk group
- Administer SIP signaling group
- Administer SIP trunk group members
- Administer FX-IP Endpoint stations

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 stations by comparing the **Maximum Off-PBX Telephones - OPS** 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 SIP endpoints to be simulated by FX-IP.

```
display system-parameters customer-options                      Page   1 of  10
                                OPTIONAL FEATURES

     G3 Version: V13
       Location: 1                             RFA System ID (SID): 1
       Platform: 6                             RFA Module ID (MID): 1

                                                                 USED
                                Platform Maximum Ports: 44000 2405
                                      Maximum Stations: 36000 1038
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 5     0
                    Maximum Off-PBX Telephones -   OPS: 200   70
                    Maximum Off-PBX Telephones - SCCAN: 0     0
```

On **Page 2** of the **OPTIONAL FEATURES** screen, verify that there is sufficient remaining capacity for SIP trunks by comparing 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 calls to be launched by FX-IP, multiplied by two, as the scenario requires two SIP trunks per simultaneous call.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

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     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 permission changes.)
```

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** contains a table listing of the audio codec types that 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   1 of   2

                          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 the following fields to allow for audio shuffling: **Intra-region IP-IP Direct Audio**, **Inter-region IP-IP Direct Audio**, and **IP Audio Hairpinning**. Increase the **'UDP Port Max'** field range from the default value to '65535'. 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 the media resources in the Avaya G650 Media Gateway.

```
change ip-network-region 2                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 2
Location:         Authoritative Domain:
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 65535
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
```

3.3. Administer IP Node Names for C-LAN and Avaya 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.

```
change node-names ip                                            Page   1 of   1
                                  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.

```
add ip-interface 1a06
                                  IP INTERFACES


                  Type: C-LAN
                  Slot: 01B04
           Code/Suffix: TN799  D
             Node Name: clan-1A06
            IP Address: 192.45 .100.147
           Subnet Mask: 255.255.255.0
       Gateway Address: 192.45 .100.1
  Enable Ethernet Port? y
        Network Region: 2
                  VLAN: n

Number of CLAN Sockets Before Warning: 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            Name: CLAN 1A06 Data Module
            Type: ethernet
            Port: 01A0617
            Link: 3

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

3.5. 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. Submit these changes.

- **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**.

Note: Although 'Direct IP-IP Audio Connections?' and 'IP Audio Hairpinning?' are displayed here with a value of 'n', the compliance tests were run with these values set to both 'y' and 'n'.

```
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
```

3.6. 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'
- **Number of Members:** '25' Note: Number of members may vary.

```
add trunk-group 10                                          Page   1 of  21
                                TRUNK GROUP

Group Number:                          Group Type: sip           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: 25
```

3.7. Administer Station Endpoints

Use the 'add station n' command, where 'n' is an available extension number. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes. Note that there is no need to set the security code, as this will be administered on the SES server.

- **Port:** Enter 'x' to indicate no hardware associated with the station.
- **Name:** Enter a descriptive name.

```
add station 11001                                              Page   1 of   5
                                     STATION

Extension: 11005                         Lock Messages? n              BCC: 0
     Type: 6408D+                        Security Code:                 TN: 1
     Port: x                           Coverage Path 1:                COR: 1
     Name: FX-IP-11001                 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: 11005
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english

          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n
```

Repeat the 'add station n' command to add the desired number of station endpoints, which is the same as the desired number of simultaneous SIP calls. For the compliance testing, twenty station endpoints were administered as shown below. When possible, use consecutive extension numbers for the stations, for ease of configuring the FX-IP.

```
list station                                                       Page   1

                                  STATIONS

Ext/          Port/   Name/                        Room/         Cv1/ COR/ Cable/
 Hunt-to       Type    Surv GK NN          Move     Data Ext      Cv2  COS  Jack
11001        X       FX-IP-11002                                     1
               6408D+                      no
11002         X       FX-IP-11002                                     1
               6408D+                      no                         1
11003         X       FX-IP-11003                                     1
               6408D+                      no                         1
11004         X       FX-IP-11004
```

For each station created above, calls to each station are routed to Avaya SES by using the 'add off-pbx-telephone station-mapping ' command.... Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Station Extension:** Station number previously created
- **Application:** Enter 'OPS' to indicate off-PBX station.
- **Phone Number:** Same digits from the **Station Extension** field.
- **Trunk Selection:** The trunk group number from **Section 3.5**.
- **Configuration Set:** An existing configuration set to be used for the off-pbx call treatment.

```
add off-pbx-telephone station-mapping                         Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 11001             OPS          -     11001               10         1
                               -                                -
List the stations listed in mapping sequence
```

Repeat the for all created SIP stations, as listed below.

```
list off-pbx-telephone station-mapping                                Page   1

                    STATION TO OFF-PBX TELEPHONE MAPPING

Station         Appl   CC  Phone Number     Config Trunk   Mapping      Calls
Extension                                   Set    Select  Mode         Allowed

11001           OPS        11001            1  /   10      both         all
11002           OPS        11002            1  /   10      both         all
11003           OPS        11003            1  /   10      both         all
11004           OPS        11004            1  /   10      both         all
11005           OPS        11005            1  /   10      both         all
```

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 Station Extension beginning with *110*.

```
change dialplan analysis                                       Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                                           Percent Full:    2

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
        0          1   fac       *         3   fac
        1          3   dac       #         3   fac
        110        5   ext
        8          1   fac
        9          1   fac
```

The incoming 5 digit number is preceded by an ‘8’. Thus ‘8110xx’ is the number called by FX-IP (note: xx is any number between ‘01’ through ‘99’). The ‘8’ is stripped off by the AAR Facilities Access Code (FAC), and the 5 digits that remained begining with ‘110xx’ are routed to an extension using AAR analysis. Below, the ‘**change aar analysis 1**’ command displays the ‘**Dialed String**’ as **110** being a ‘**Total Length**’ of ‘**5’**, and the ‘**Call Type**’ is an ‘**ext**’ (extension). The call is routed to ‘**Route Pattern**’ number ‘**10**’ and the ‘**Call Type**’ of ‘**aar**’.

```
change aar analysis 1                                   Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                                    Percent Full:    2

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    110                   5    5     10       aar          n
    2                     7    7     999      aar          n
```

Lastly the '**change route-pattern 10**' command indicates the '**Grp No**' (or Trunk Group) that will be used to route the out bound traffic, and in this case is '**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                              QSIG
                             Dgts                                     Intw
 1: 10   0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                          none
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

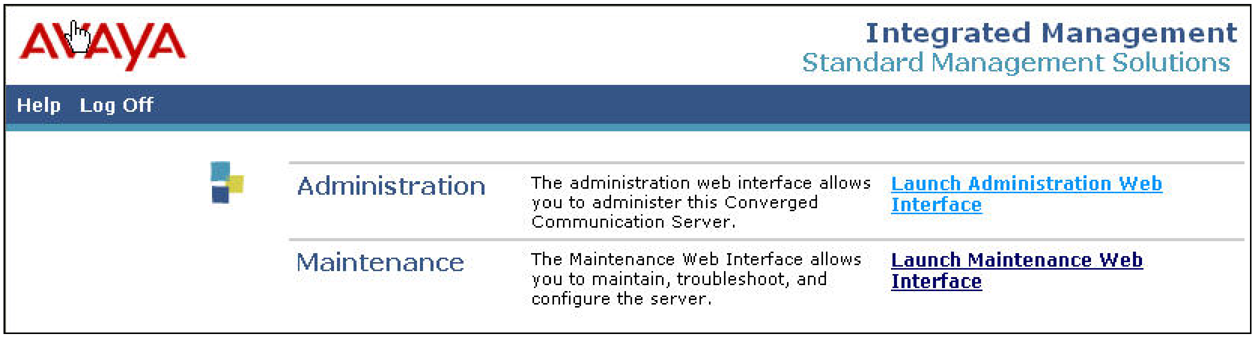
4. Configure Avaya SIP Enablement Services

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

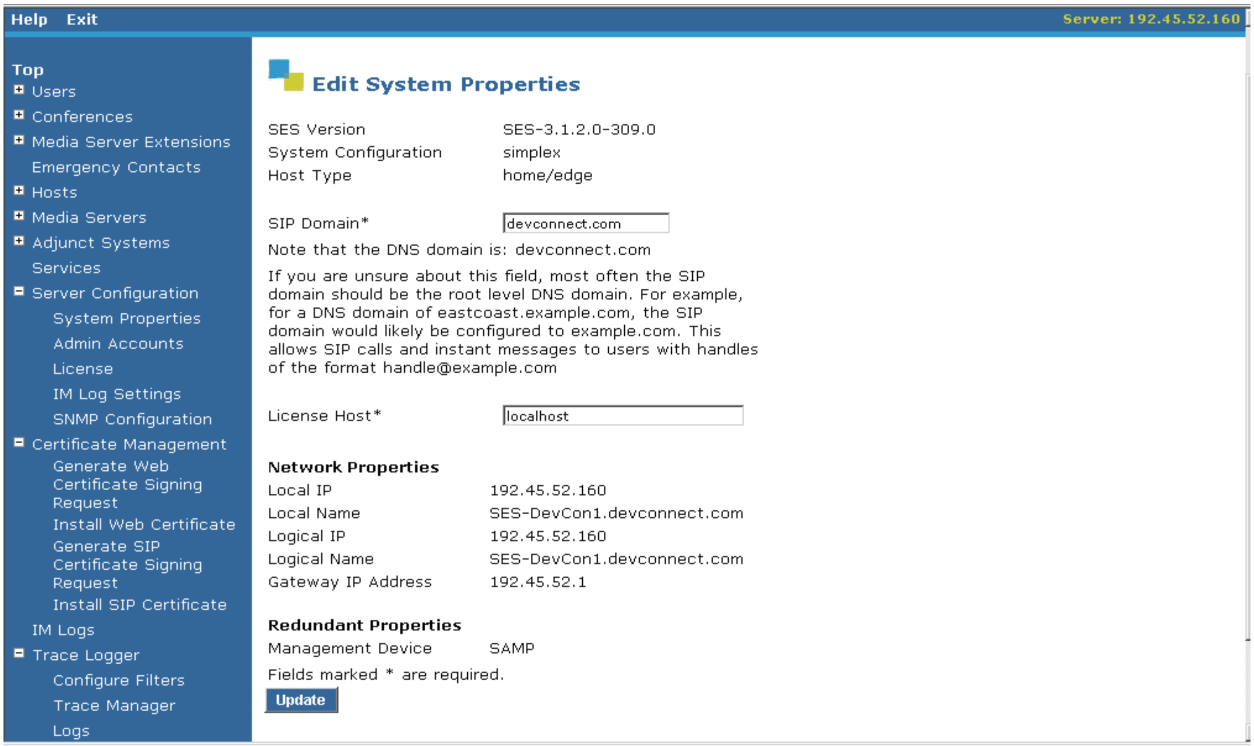
4.1. Obtain SIP Domain and Host

Access the 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 Avaya 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.

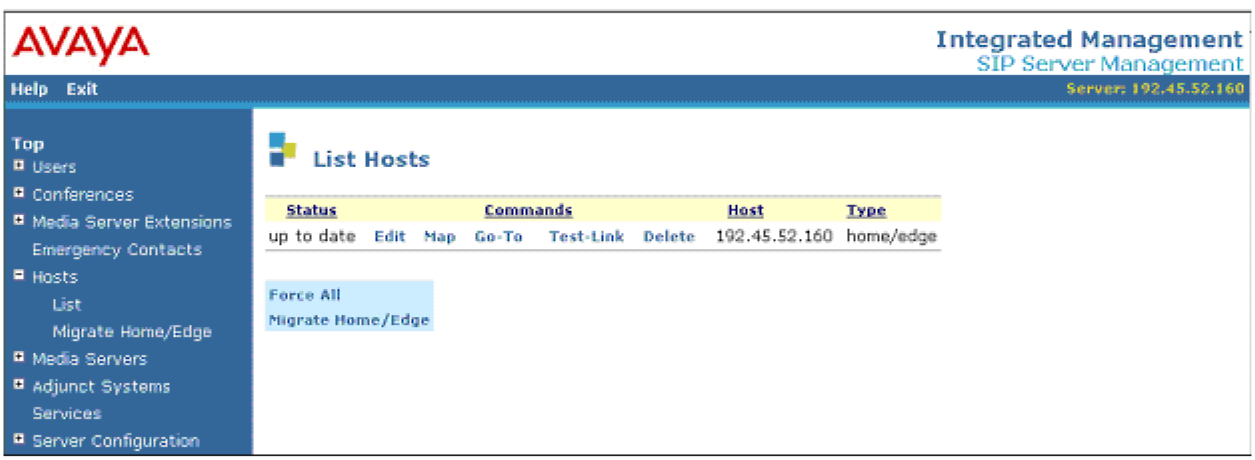


The **Top** screen is displayed next. If this is the initial setup of the Avaya SES server, then follow the SIP Enablement Services Installation and Administration Guide [2] to administer the SIP domain and host. These Application Notes assume the Avaya SES server has already been configured with the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

Select **Server Configuration** > **System Properties** from the Left Panel to display the **Edit System Properties** screen below. Make a note of the value in the **SIP Domain** field, in this case 'devconnect.com', as this may be used later to configure FX-IP.

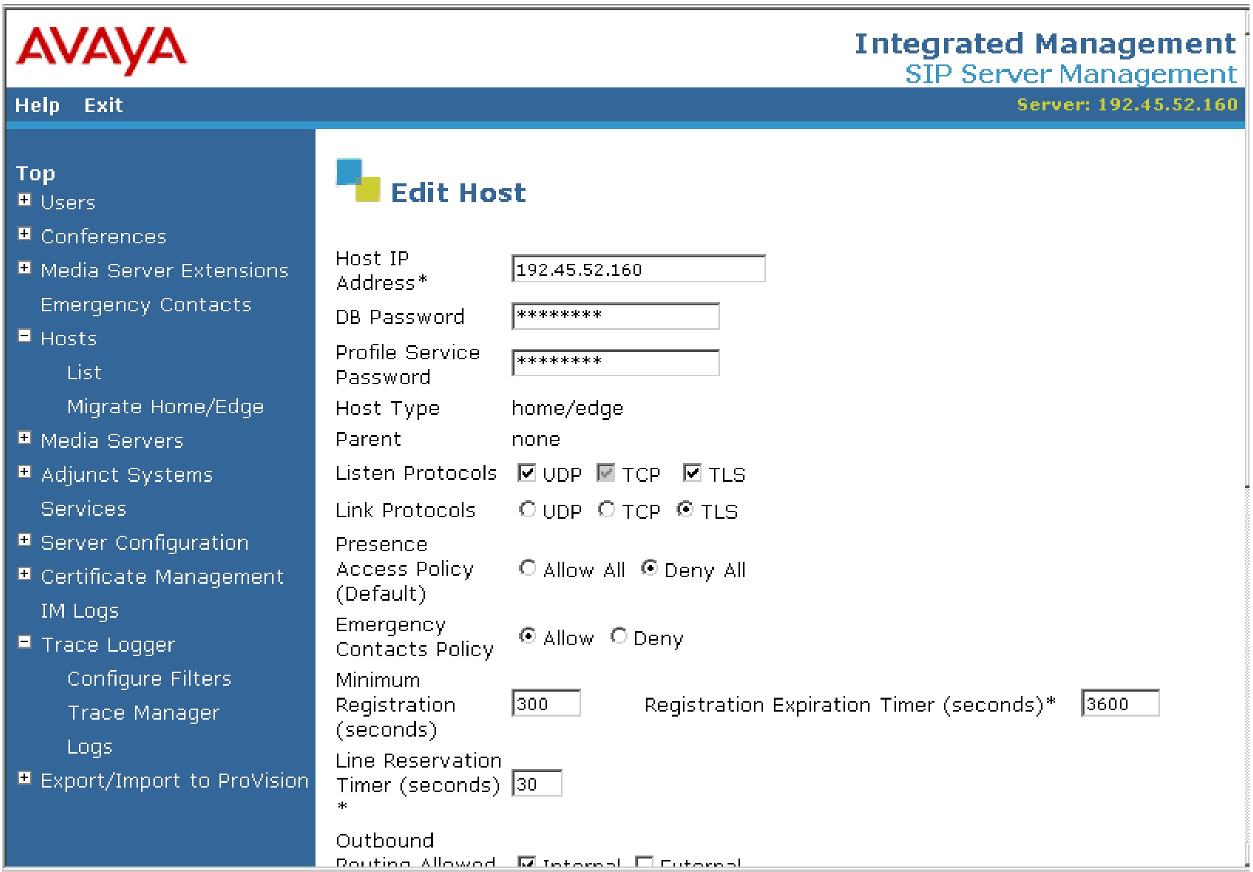


Select **Hosts** from the Left Panel to display the **List Hosts** screen. Click on the **Edit** button for each host.



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. For the compliance testing, only one host is administered as both the home and edge server, as indicated by the **Host value** 192.45.52.160 and **Type** value '**home/edge**.

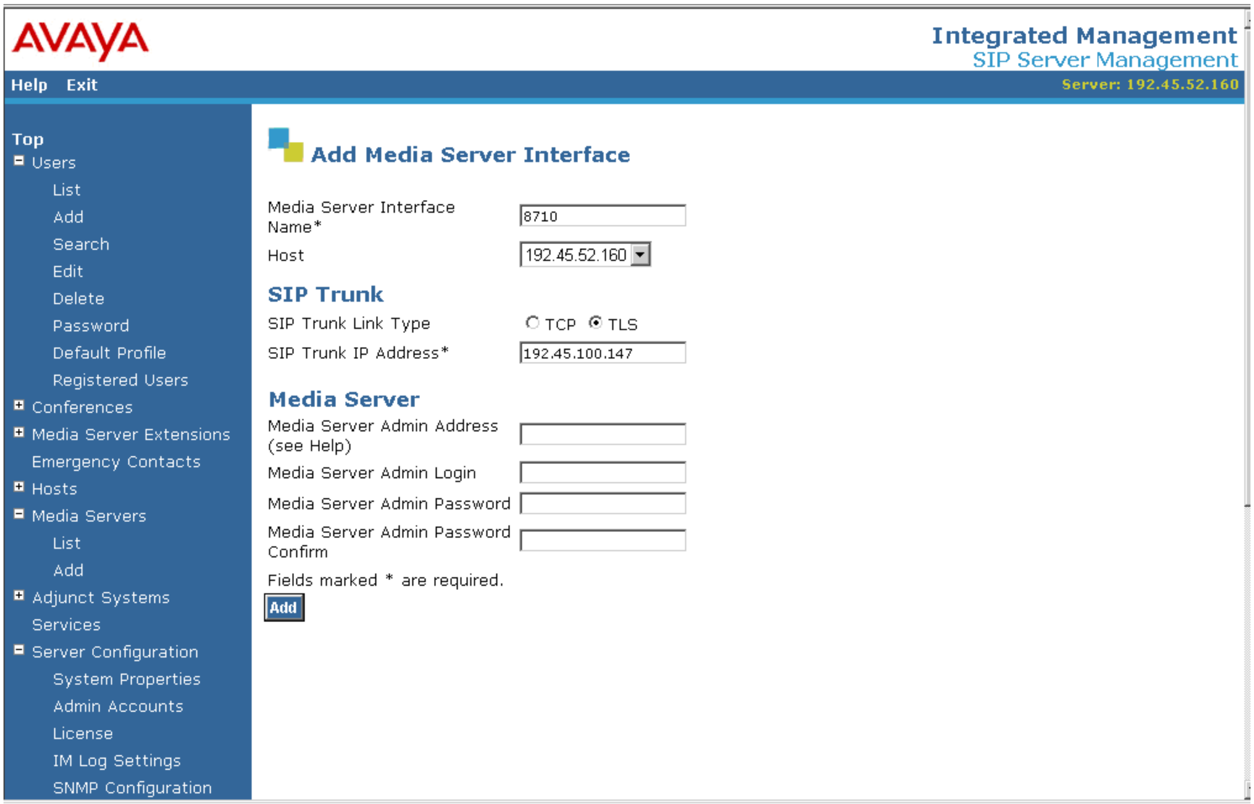
The host name of this home server is '192.45.52.160', indicated in the **Host IP Address** field.



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.

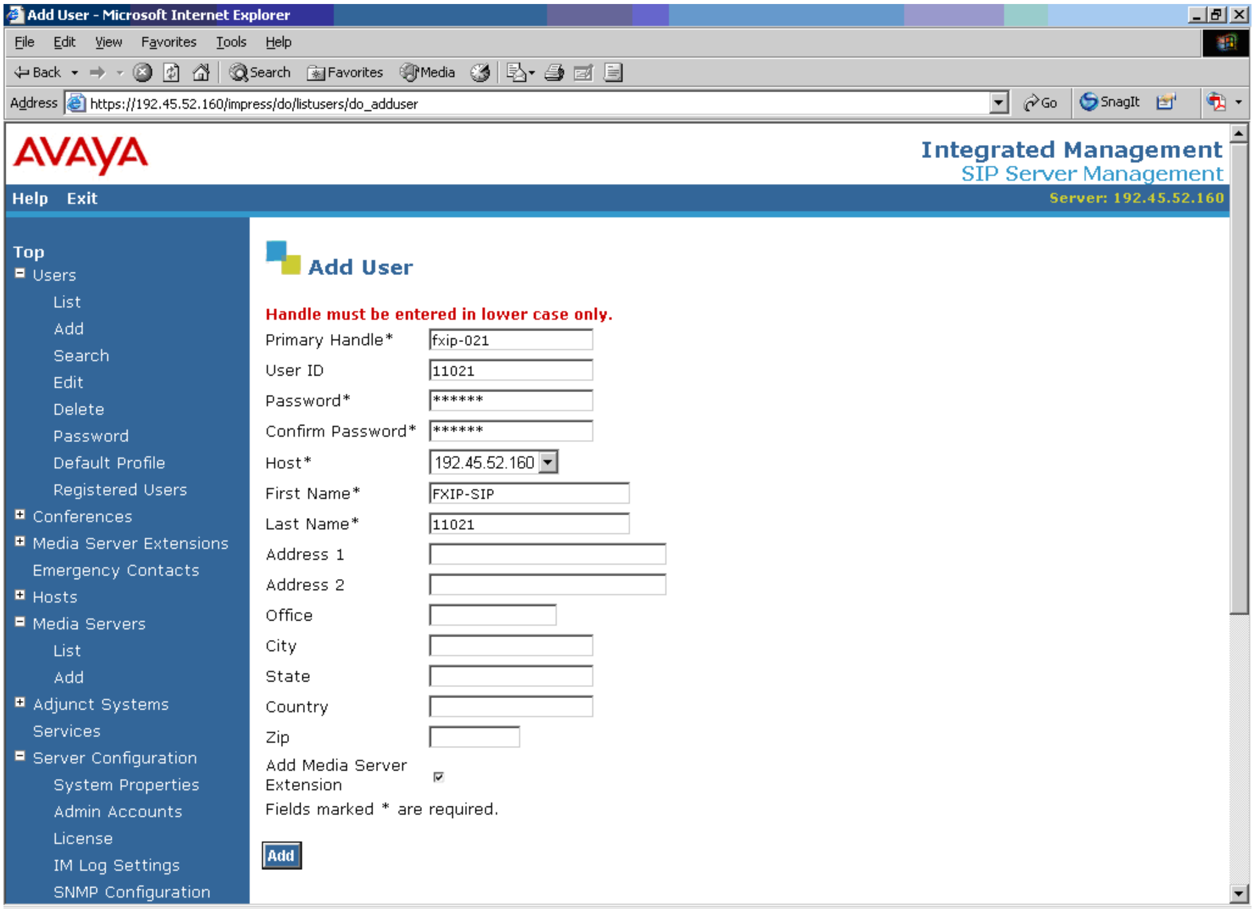
- **Media Server Interface:** A descriptive name.
- **Host:** Select the host name of the home server from **Section 4.1**.
- **SIP Trunk IP Address:** Enter the C-LAN IP address from **Section 3.3**.



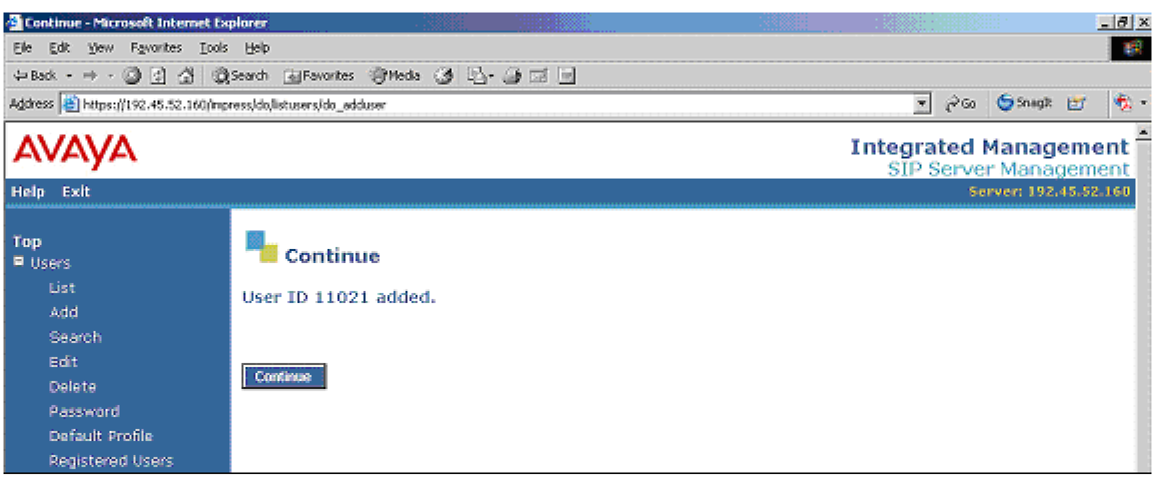
4.3. Administer Users

Select **Users** > **Add** from the Left Panel to display the **Add User** screen. Enter the following values for the specified fields to add users. Click on **Add** in the bottom of the screen to submit these changes.

- **Primary Handle:** The extension of the SIP station from **Section 3.8**. **Note**: The value placed in this field does not have to be the extension number but normally is used.
- **Password:** Enter a desired password, in this case '123456".
- **Confirm Password:** Re-enter the same password, in this case '123456".
- **Host:** Select the host name of the home server from **Section 4.1**.
- **First Name**: A descriptive first name.
- **Last Name:** A descriptive last name.
- **Add Media Server Extension:** Check the box.

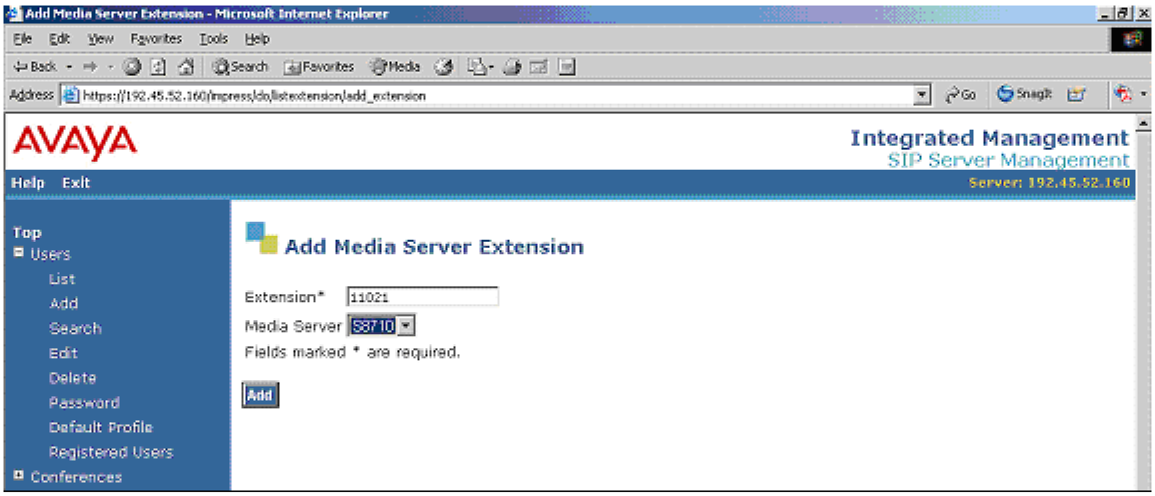


The **Continue** screen is displayed next. Click on the **Continue** button in the bottom of the screen.

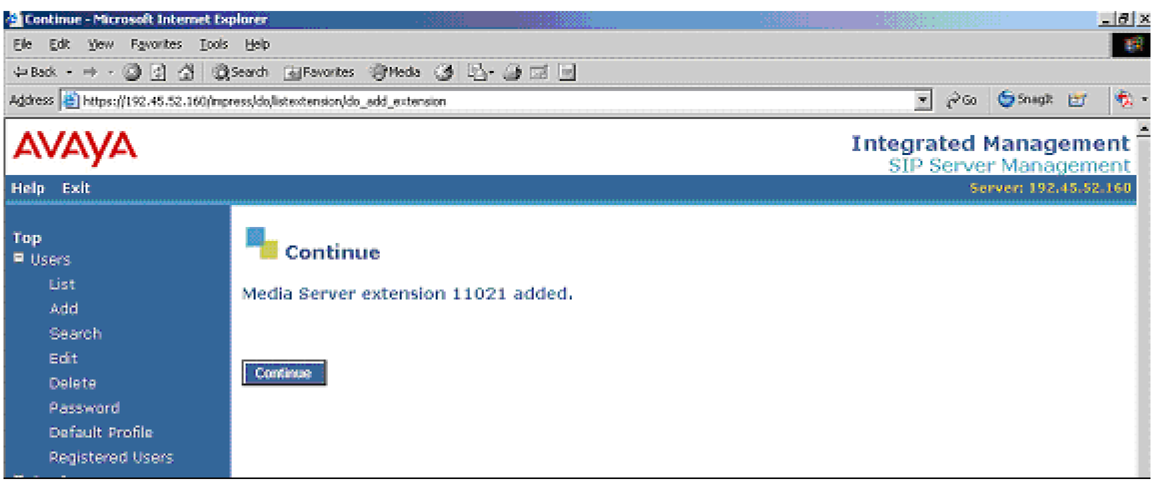


The **Add Media Server Extension** screen is displayed next. This screen is used to associate a user with a media server extension. Enter the following values for the specified fields. Click on **Add** in the bottom of the screen to submit these changes.

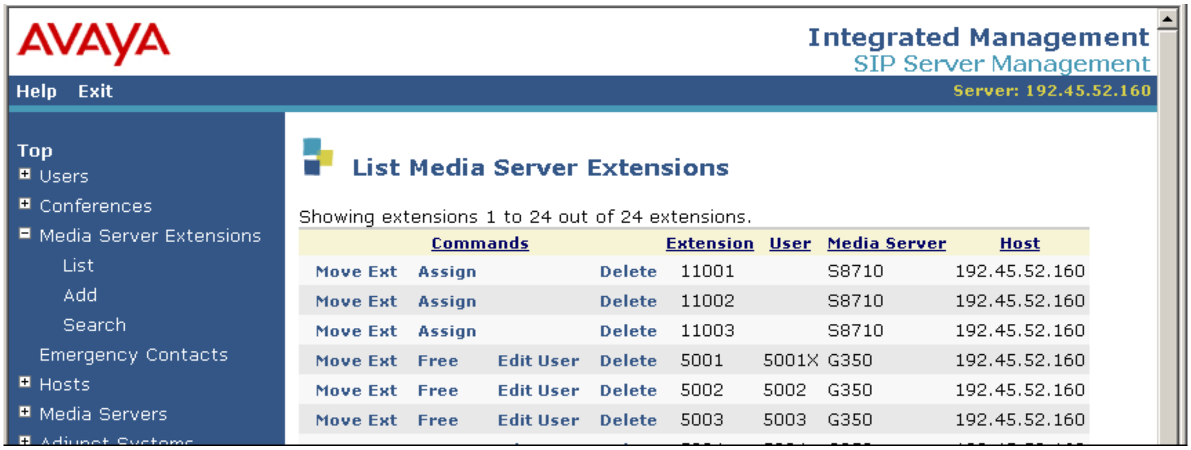
- **Extension:** The extension of the SIP station from **Section 3.8**.
- **Media Server:** Select the name of the media server interface from **Section 4.2**.



The **Continue** screen is displayed next. Click on the **Continue** button in the bottom of the screen.



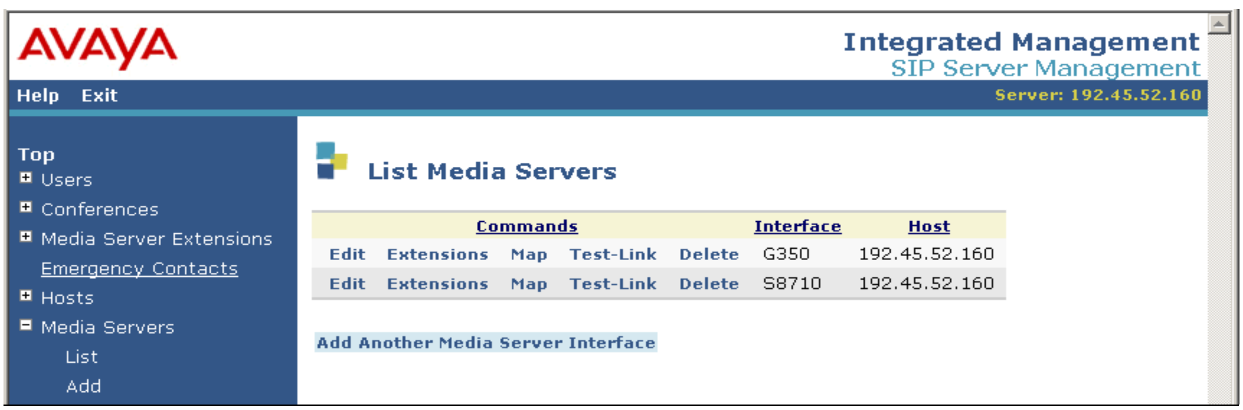
Repeat these procedures to add the desired number of SIP users and associated media server extensions. When possible, use consecutive numbers for the SIP users, for ease of configuring FX-IP later on.



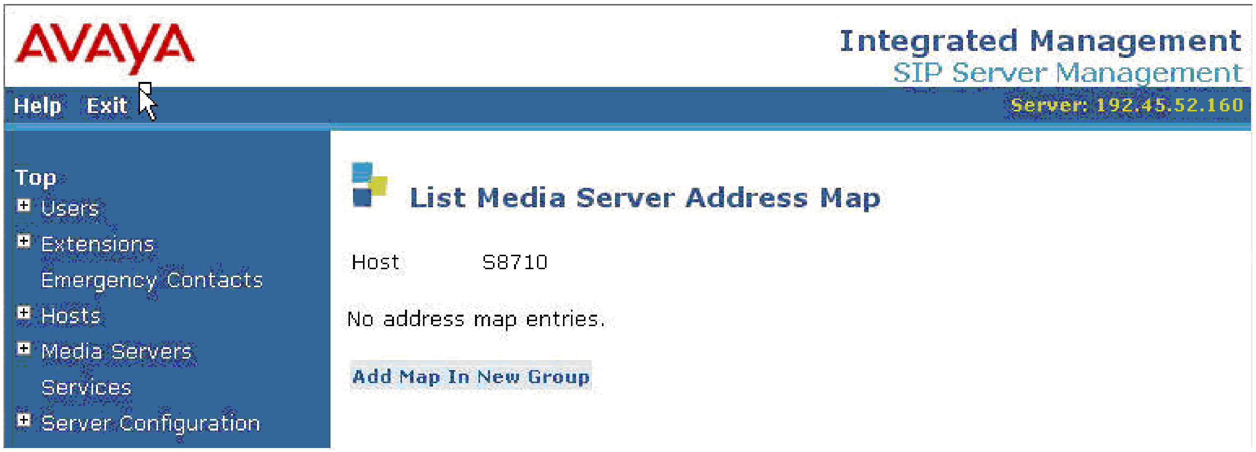
For the compliance testing, twenty SIP users and extensions were administered.

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'.



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.

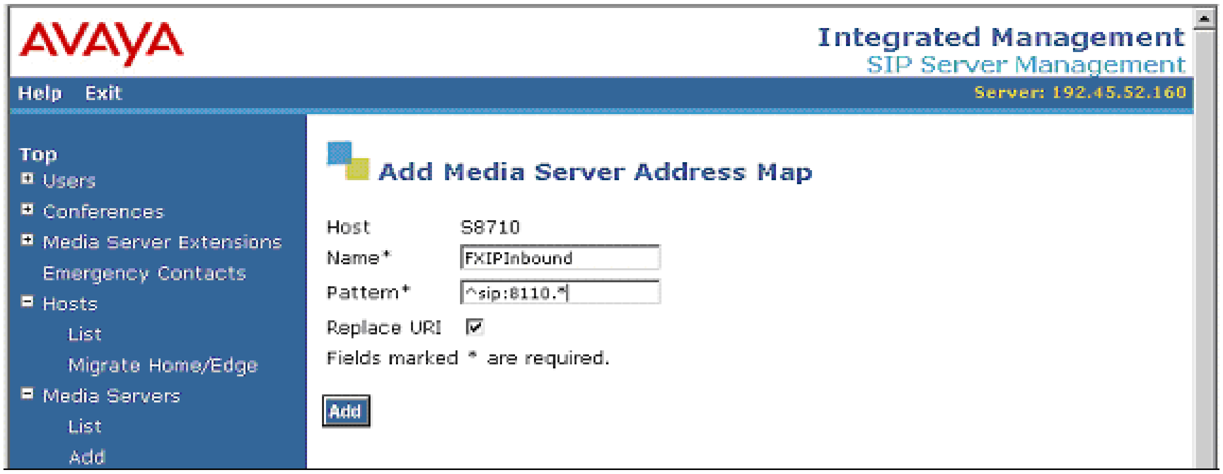
For the **Name** field, enter a descriptive name to denote the routing. For the compliance testing, incoming SIP calls to extensions 8110xx (xx denotes numbers 01 through 99) are to be routed to media server 'S8710'. Therefore, the value '8110xx' is entered to denote the name of this media server address mapping.

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:

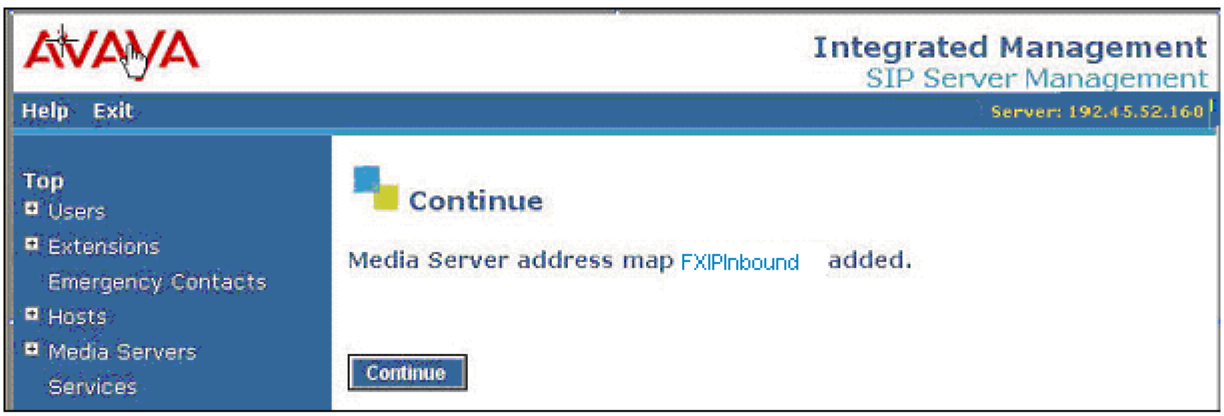
- **^sip:** Match to any SIP INVITE message with 'sip:' at the beginning of the URI.
- **8110** Matching the inbound extension digits beginning with '8110'.
- **.*** Match the last two digits of the extension to any digits.

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].

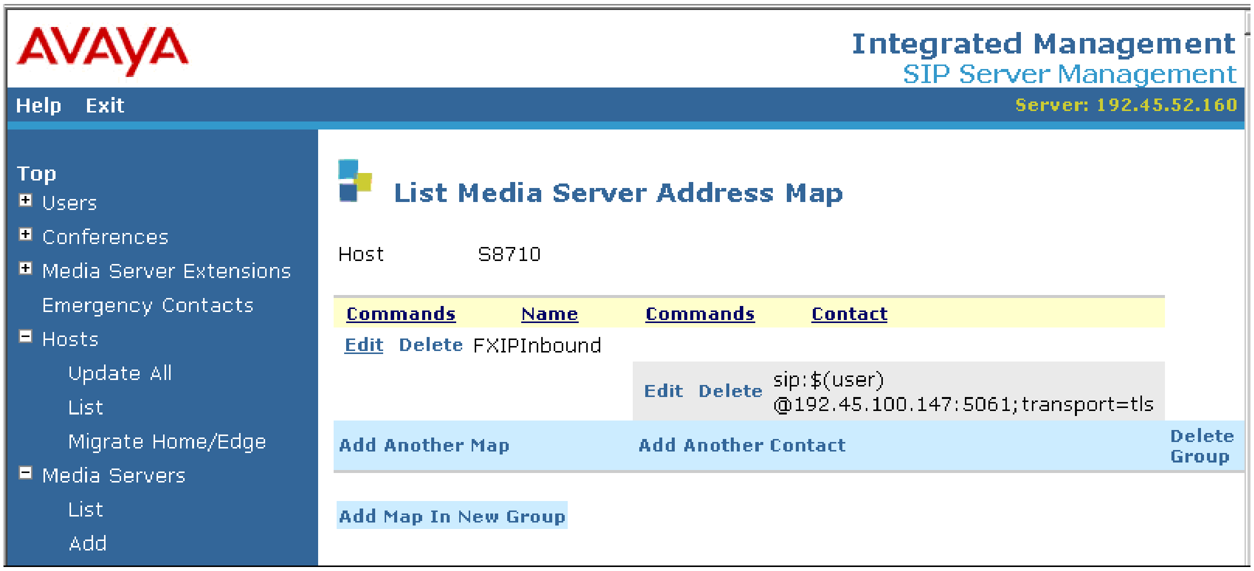
Verify the **Replace URI** field is checked, to enable SES to replace the URI in the incoming SIP INVITE messages with C-LAN contact information, in order to reach Avaya Communication Manager. Click on **Add** in the bottom of the screen.



The **Continue** screen is displayed next. Click on the **Continue** button.



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.



4.5. Administer Trusted Host

Administer the FX-IP as a trusted host so that the SIP 'Bye' messages from FX-IP system 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-DevCon1> 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 Empirix FX-IP

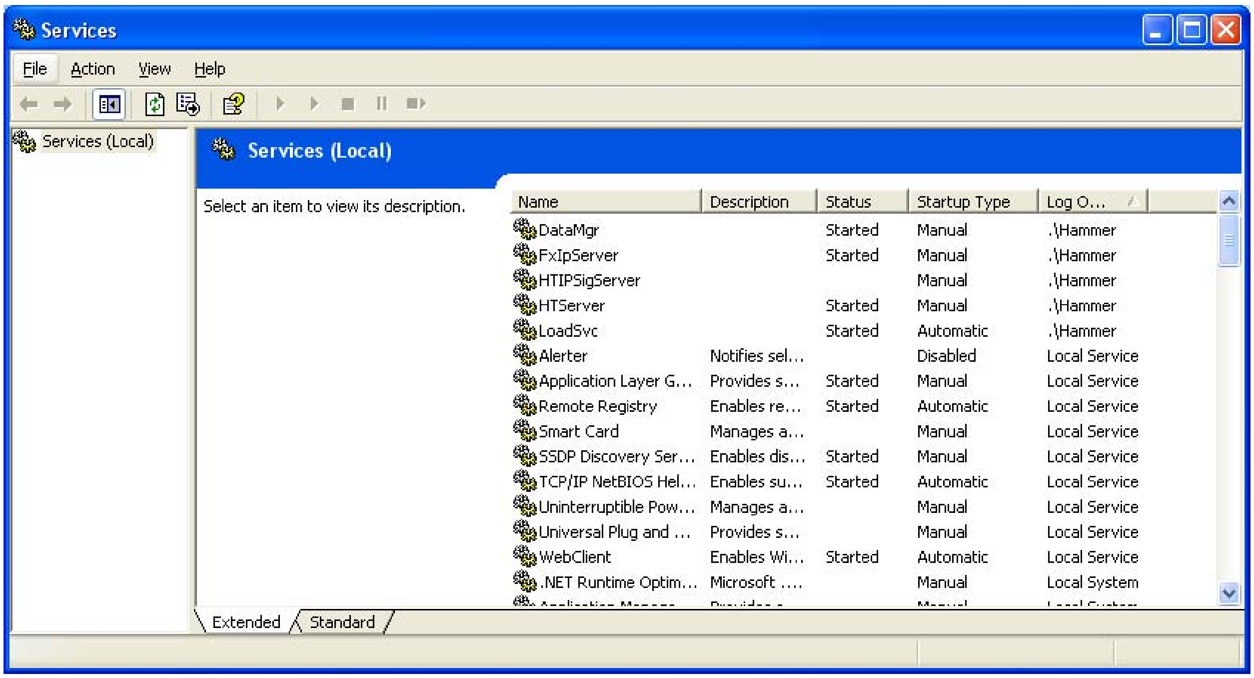
The FX-IP Hammer Configurator is the application used to configure the FX-IP Hammer system. The FX-IP Hammer Configurator is a GUI client application that can configure a local or remote FX-IP Hammer Configurator server via a network connection.

Note: The Empirix FX-IP configuration steps displayed in this document may not truly reflect all the procedural steps required to interoperate with Empirix FX-IP, but should be used as a guideline.

5.1. Empirix Hammer Configurator

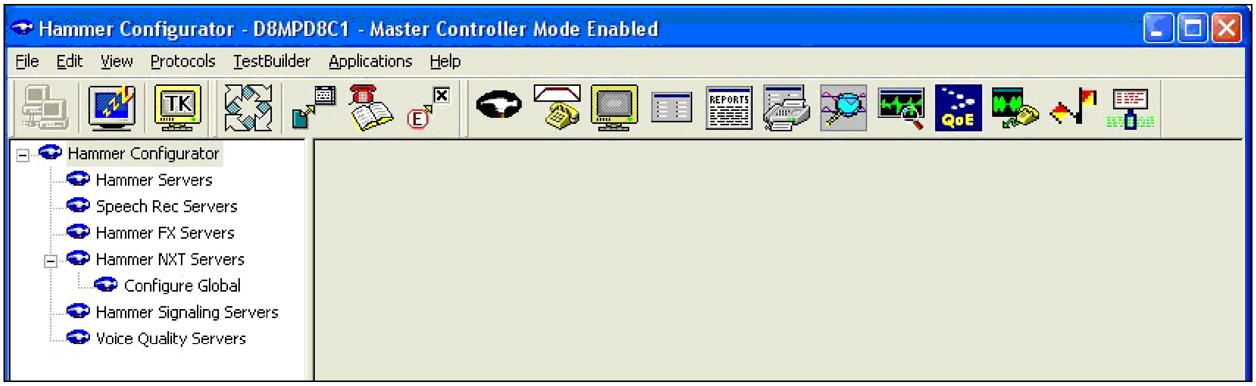
The procedure in this section is used to configure the originating and terminating SIP interfaces on FX-IP to interoperate with Avaya Communication Manager. Use the following steps to configure the FX-IP.

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

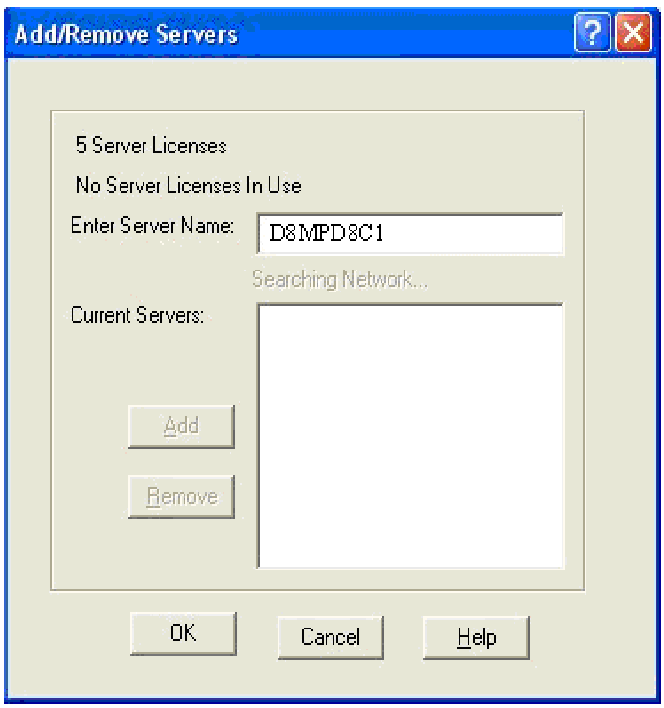


1. Start the 'Hammer Configurator' from the 'Start Menu' by selecting Programs → Hammer → **Hammer Configurator**.

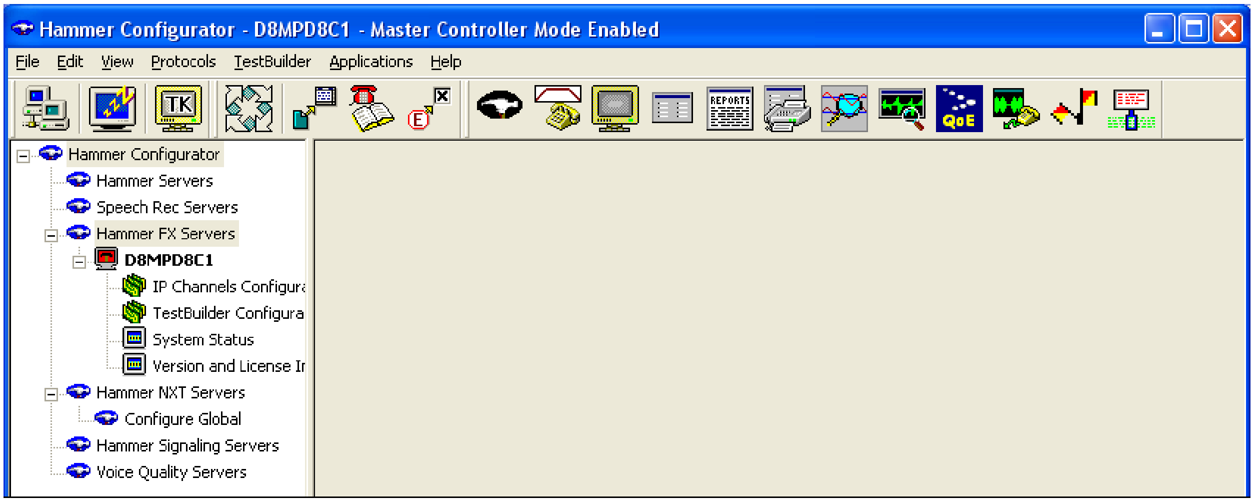
2. Highlight **'Hammer FX Server**' on the Left Panel, and then click on 'Edit'. Under the 'Edit' pull-down menu, select '**Add/Remove Servers**'.



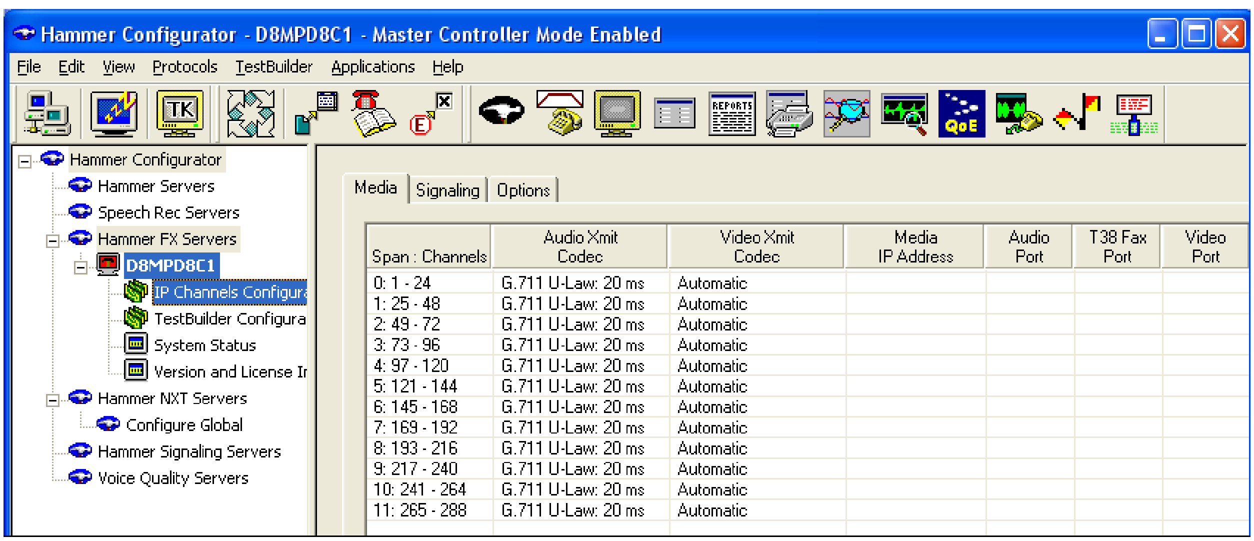
3. 'Enter Server Name'. This is where the Empirix FX-IP software resides. In this case it's 'D8MPD8C1'. Names will vary. Click 'OK'.



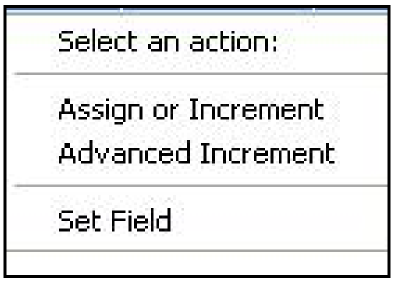
4. As a result of entering the Server name, the 'Hammer FX Server' entries should be added as below.

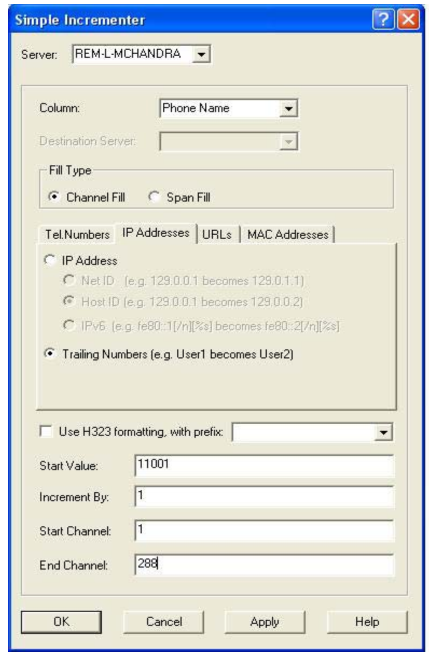


5. On the Left Panel, click on **'IP Channels Configuration'**. The Right Panel opens up displaying the 'Media' tab. Use the 'Media' tab to configure the parameters for sending and receiving media on the channels. To add/change any parameter on each row (or multiple rows by holding down the "Shift" key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



When a row on any of the 17 configurable columns is 'right clicked', the configurable information can be selected from the list provided or, it can be incrementally inputted when the action dialog boxes displayed below that come into view. Multiple channels (rows) can be selected by holding the 'Shift' key down while selecting the additional channels.





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. This field cannot be modified.	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 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 terminal 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 20 for each channel.

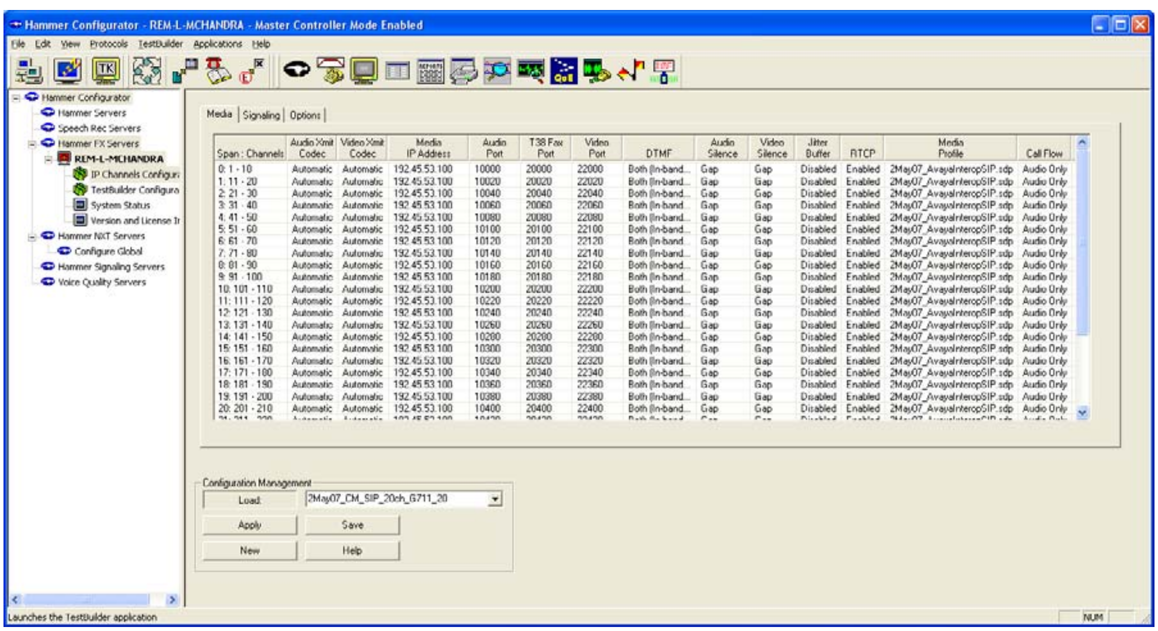
T.38 Fax Port	Specifies the UDP port for T.38 fax (SIP and MGCP protocols only). This option is only visible if licensed for T.38 Fax.	Start at 20000 incremented by 20 for each channel. Not used.
Video Port	Specifies the UDP port number for the video media. This is the port number a channel sends video from and receives video on. This option is only visible if licensed for video.	Start at 22000 incremented by 20 for each channel. Not used.
DTMF	Specifies the DTMF transmission method.	Both Directions (Inband)
Audio Silence	Specifies the action of an audio stream when it is idle.	GAP
Video Silence	Specifies the action of a video stream when it is idle. This column is only visible if licensed for video.	GAP
Jitter Buffer	Enable or disable a jitter buffer.	Disabled
RTCP	Enable or disable transmission of RTCP packets with media.	Enabled
Media Profile	Specifies the current media profile. The media profile should match the media that is configured on the Media Tab.	The name of the .SDP profile under test.
Destination Media IP	Specifies the IP address the media is sent to when using the Clear Channel protocol. This option is only visible when one or more spans are configured for Clear Channel.	Not Used
Destination Audio Port	Specifies the RTP/UDP port to which the media is sent when using the Clear Channel protocol. This option is only visible when one or more spans are configured for Clear Channel.	Not Used
Destination T.38 Port	Specifies the UDP port to which the fax is sent when using the Clear Channel protocol. This option is only visible if licensed for T.38 Fax and one or more spans are configured for Clear Channel.	Not Used

Call Flow	Specifies the order in which media profile capabilities are declared for a multimedia session using a single media stream. This option is only visible if licensed for video or for T.38 Fax.	Audio only

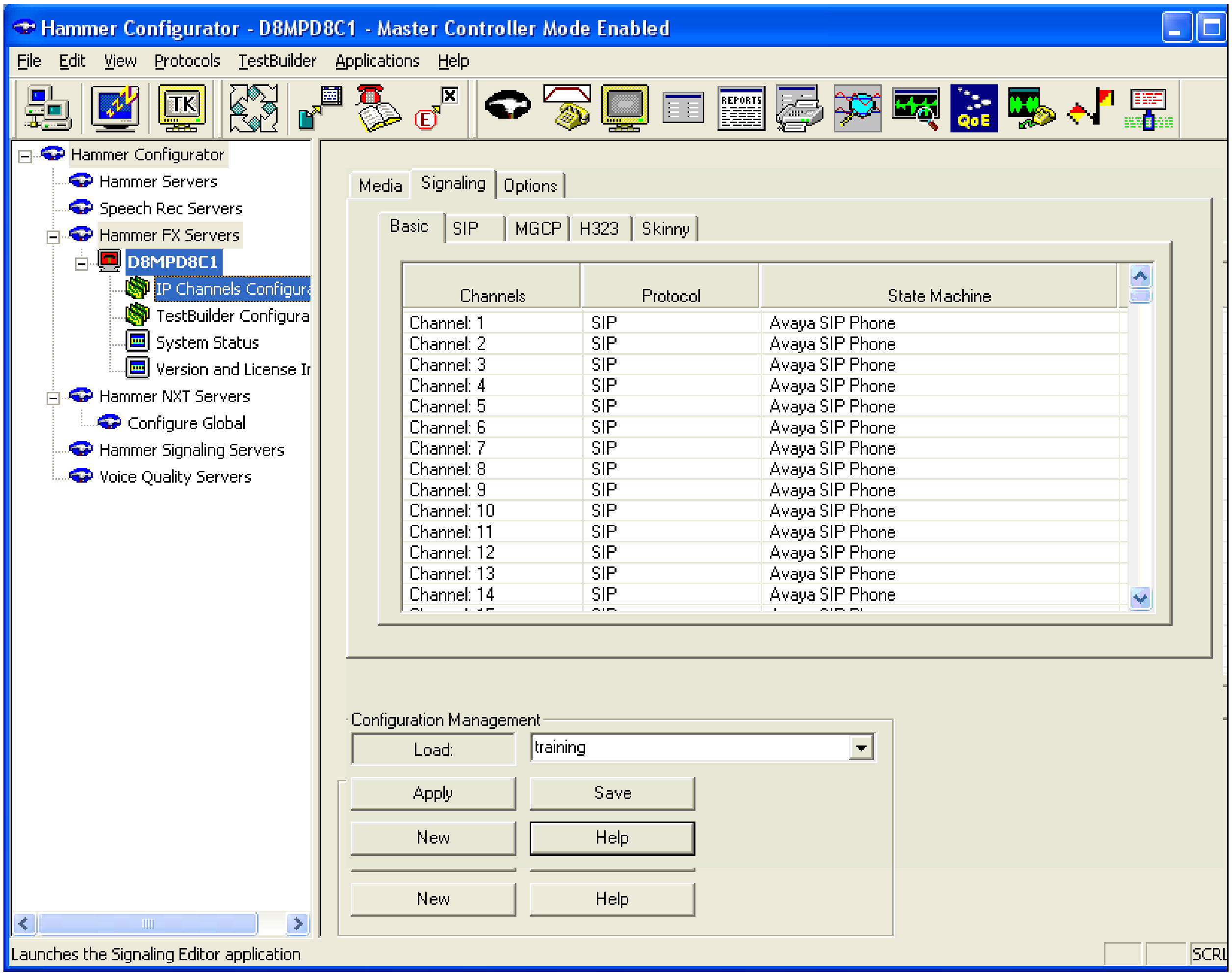
6. Note: The codec changes were applied for SIP Endpoint testing via the 'Media Profile Files' files indicated below. The files contain the configuration information on how the FX-IP is setup.

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 Endpoint test as indicated in the snapshot below.



7. Configure the protocol and signaling on each channel by clicking the 'Signaling' tab (or multiple rows by holding down the "Shift' key while selecting).

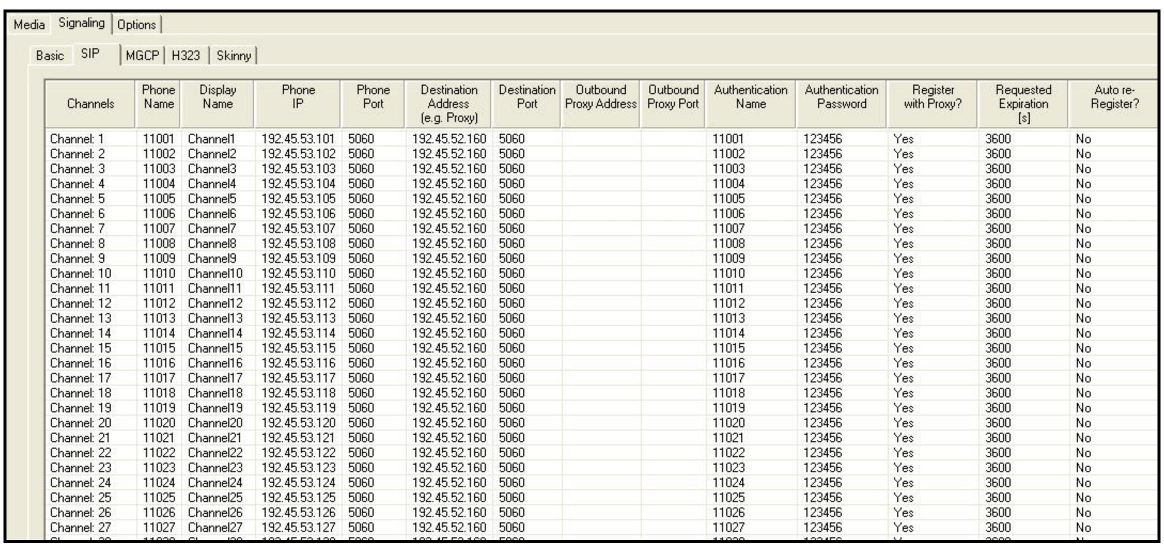


- Explanation of the Columns under the 'Signaling → Basic' tabs are expressed in the table below:

Parameters	Descriptions	Test Value
Channels	Displays identifiers for the IP channels in the system, one channel per row. Cannot modify this field.	288 channels. Only 20 used.
Protocol	Select the protocol that provides IP signaling. You must assign the protocol within span boundaries; a protocol cannot be assigned on individual channels. To select the protocol, right-click and select one of the following protocols from the drop-down list:	SIP
State Machine	Select the state machine that defines the behavior of an emulated Trunk. Right-click and select a name from the drop-down menu, or select Set Field and specify a state machine name in the text .box.	Empirix Modified to accommodate IP Direct Media (shuffling). Otherwise, select Avaya SIP Phone. See test results, **Section 6.2**.

8. The **'**Signaling → SIP**'** tab is used to configure the SIP parameters. The following two screen shots display the parameters configured for the SIP Endpoint test. Note: Each of the 'Phone IP' addresses is different.

- 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	192.45.53.101	5060	192.45.52.160	5060			11001	123456	Yes	3600	No
Channel: 2	11002	Channel2	192.45.53.102	5060	192.45.52.160	5060			11002	123456	Yes	3600	No
Channel: 3	11003	Channel3	192.45.53.103	5060	192.45.52.160	5060			11003	123456	Yes	3600	No
Channel: 4	11004	Channel4	192.45.53.104	5060	192.45.52.160	5060			11004	123456	Yes	3600	No
Channel: 5	11005	Channel5	192.45.53.105	5060	192.45.52.160	5060			11005	123456	Yes	3600	No
Channel: 6	11006	Channel6	192.45.53.106	5060	192.45.52.160	5060			11006	123456	Yes	3600	No
Channel: 7	11007	Channel7	192.45.53.107	5060	192.45.52.160	5060			11007	123456	Yes	3600	No
Channel: 8	11008	Channel8	192.45.53.108	5060	192.45.52.160	5060			11008	123456	Yes	3600	No
Channel: 9	11009	Channel9	192.45.53.109	5060	192.45.52.160	5060			11009	123456	Yes	3600	No
Channel: 10	11010	Channel10	192.45.53.110	5060	192.45.52.160	5060			11010	123456	Yes	3600	No
Channel: 11	11011	Channel11	192.45.53.111	5060	192.45.52.160	5060			11011	123456	Yes	3600	No
Channel: 12	11012	Channel12	192.45.53.112	5060	192.45.52.160	5060			11012	123456	Yes	3600	No
Channel: 13	11013	Channel13	192.45.53.113	5060	192.45.52.160	5060			11013	123456	Yes	3600	No
Channel: 14	11014	Channel14	192.45.53.114	5060	192.45.52.160	5060			11014	123456	Yes	3600	No
Channel: 15	11015	Channel15	192.45.53.115	5060	192.45.52.160	5060			11015	123456	Yes	3600	No
Channel: 16	11016	Channel16	192.45.53.116	5060	192.45.52.160	5060			11016	123456	Yes	3600	No
Channel: 17	11017	Channel17	192.45.53.117	5060	192.45.52.160	5060			11017	123456	Yes	3600	No
Channel: 18	11018	Channel18	192.45.53.118	5060	192.45.52.160	5060			11018	123456	Yes	3600	No
Channel: 19	11019	Channel19	192.45.53.119	5060	192.45.52.160	5060			11019	123456	Yes	3600	No
Channel: 20	11020	Channel20	192.45.53.120	5060	192.45.52.160	5060			11020	123456	Yes	3600	No
Channel: 21	11021	Channel21	192.45.53.121	5060	192.45.52.160	5060			11021	123456	Yes	3600	No
Channel: 22	11022	Channel22	192.45.53.122	5060	192.45.52.160	5060			11022	123456	Yes	3600	No
Channel: 23	11023	Channel23	192.45.53.123	5060	192.45.52.160	5060			11023	123456	Yes	3600	No
Channel: 24	11024	Channel24	192.45.53.124	5060	192.45.52.160	5060			11024	123456	Yes	3600	No
Channel: 25	11025	Channel25	192.45.53.125	5060	192.45.52.160	5060			11025	123456	Yes	3600	No
Channel: 26	11026	Channel26	192.45.53.126	5060	192.45.52.160	5060			11026	123456	Yes	3600	No
Channel: 27	11027	Channel27	192.45.53.127	5060	192.45.52.160	5060			11027	123456	Yes	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

NOTE: The ‘Use Shuffling’ column has been customized by Empirix and it’s for enabling / disabling the use of the shuffling feature in Avaya Communication Manager.

- 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. 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 subnet address as the 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 messages.
Destination Address	Yes	IP address or domain name of each channel's 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 Port	Yes	The SIP UDP port in-use at each Proxy IP address for sending and listening for SIP messages.
Authentication Name	No	Name used in the username/password authentication scheme.
Authentication Password	No	Password used in the username/password authentication scheme.
Register with Proxy?	No	Select whether or not a channel 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

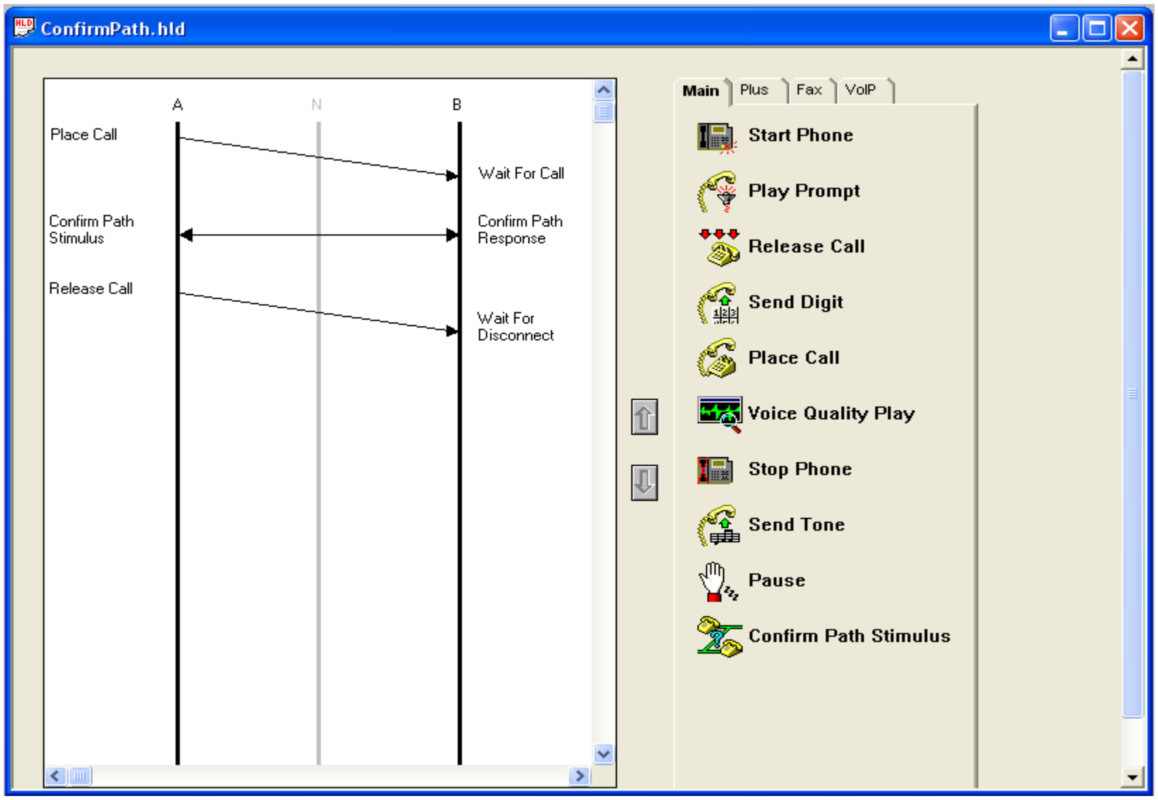
5.2. Empirix Test Builder

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

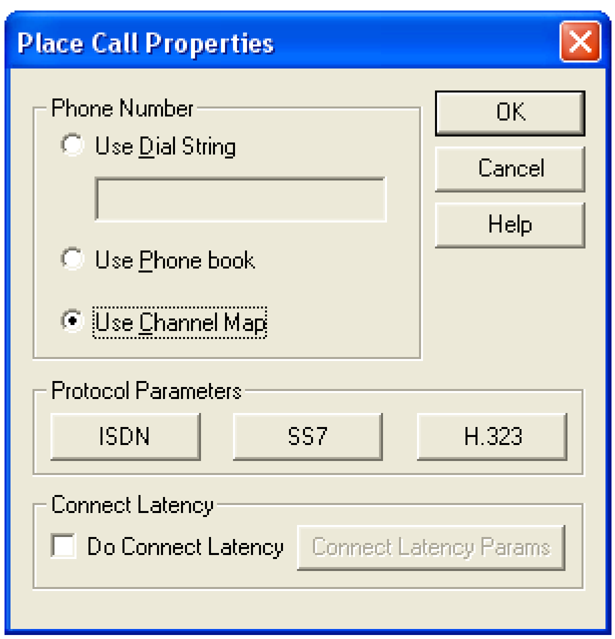
TestBuilder provides the following:

- Two interfaces for creating tests. 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 the tests are run.
- 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 enables a simulated real world operating conditions. 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 selected.

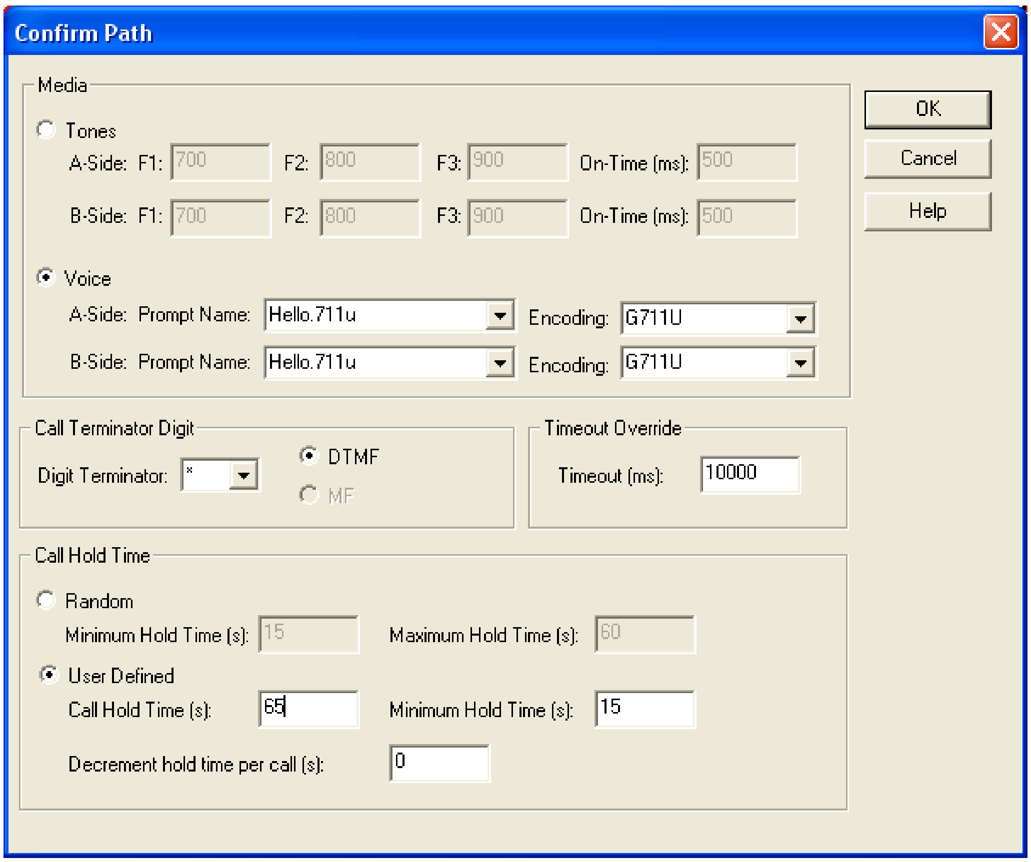
Use a predefined test case with bidirectional voice path confirmation from A to B and vice versa. This test case is executed throughout the interoperability testing with Avaya Communication Manager.



A 'Right Click' on the 'Place Call' state brings up a dialog box that includes 'Properties'. Clicking on the 'Properties' displays the following screen. The method used to assigning telephone number strings can be configured within the 'Place Call' properties.



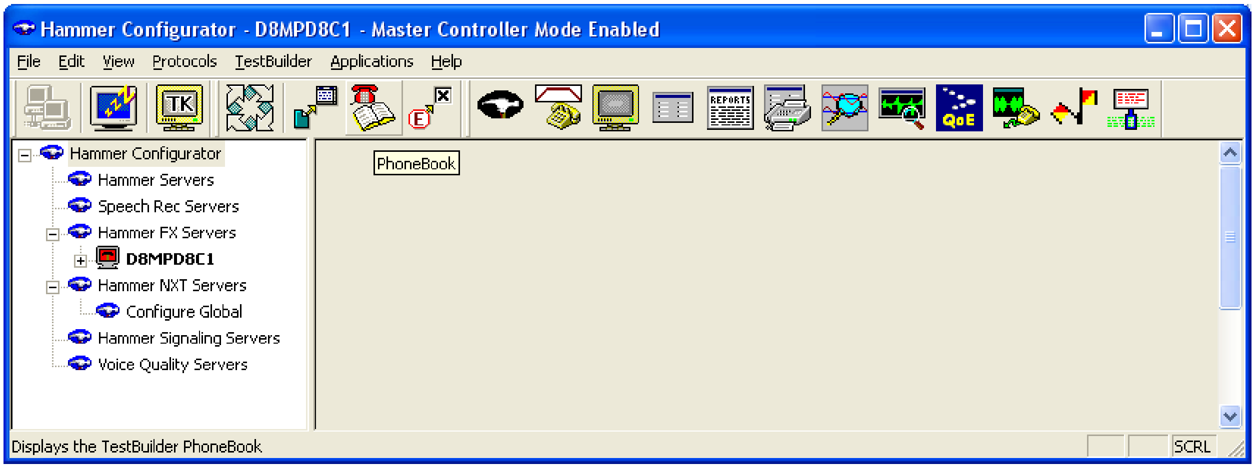
A ‘Right Click’ on the ’Confirm Path’ state brings up a dialog box that includes ‘Properties’. Clicking on the ‘Properties’ displays the following screen. The ‘Confirm Path’ state verifies that a bidirectional path is open between the A-Side and B-Side along with various user settable parameters as below.



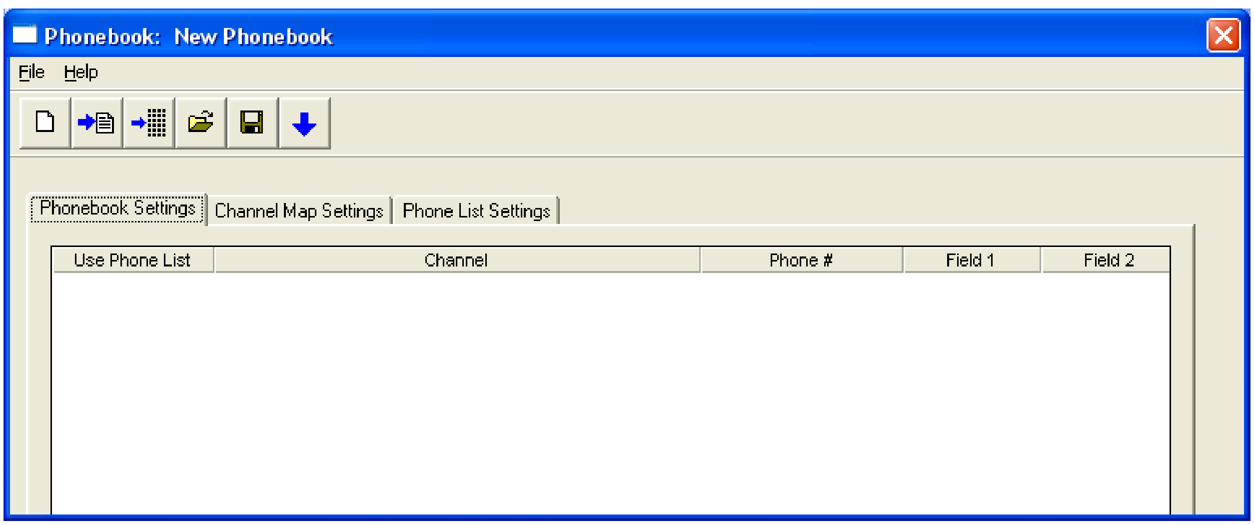
5.3. Phone Book Setup

Use the PhoneBook Settings tab to enter phone numbers to call. Enter numbers on a channel or span basis. If one number is entered 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 [illegible]ify the Master Controller mode is enabled as indicated on the top line of ‘Hammer[illegible]onfigurator’ application window.[ehope1]



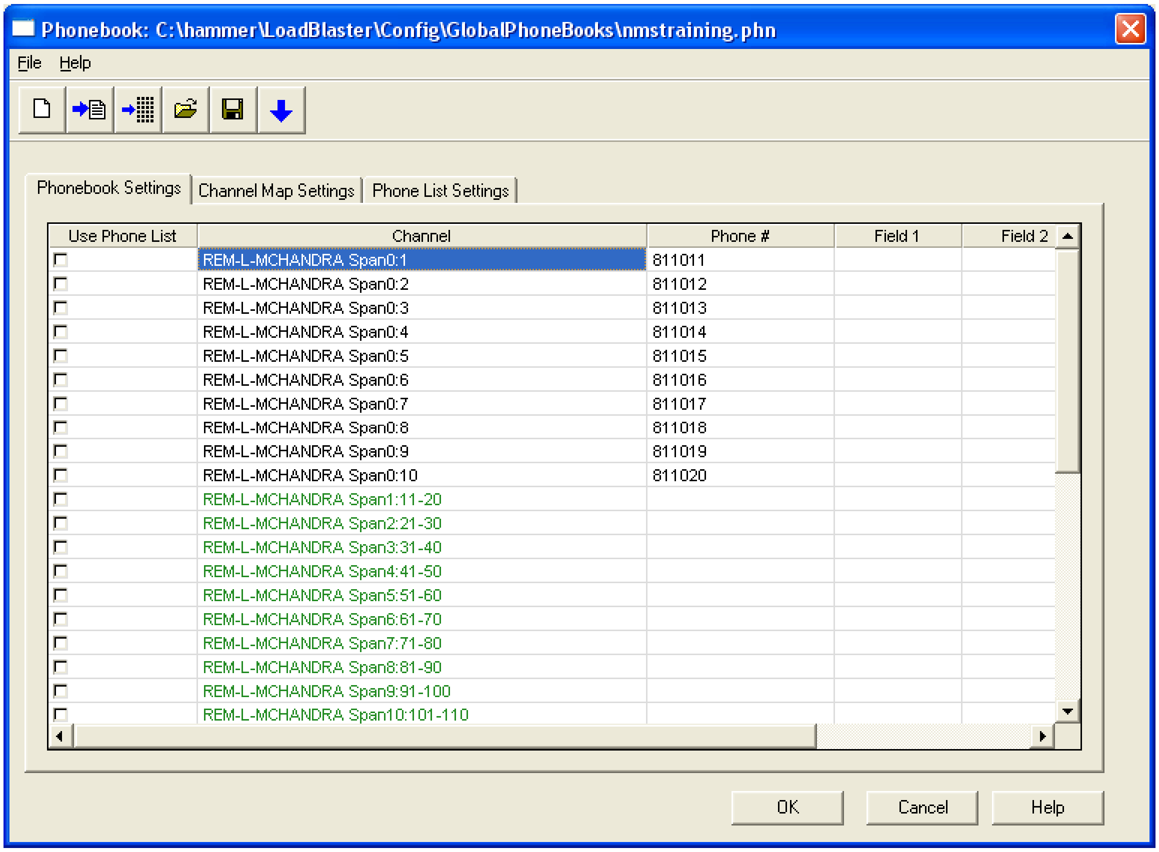
2. Open the PhoneBook e2]. by selecting TestBuilder → PhoneBook on the Hammer Configurator Menu. T last saved PhoneBook opens. If the last saved PhoneBook cannot be found, a new blank PhoneBook opens.



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.

The Phone book used for SIP Trunk Testing is shown below.



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.

Select File → Exit to close the PhoneBook Suite window.

6. Interoperability Compliance Testing

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

- Generation of moderate SIP endpoint load to Avaya Communication Manager via the trunk interface to Avaya SES, and back out to the SIP endpoints 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.

6.1. General Test Approach

The feature test cases were conducted by using FX-IP to originate and terminate calls from registered 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.

6.2. 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.

7. Verification Steps

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

7.1. Verify Avaya Communication Manager

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 trunk group by using the 'status trunk n' command, where 'n' is the trunk group number administered in **Section 3.5**. Verify that the appropriate trunk channels are in the 'in-service/active' state as shown below. Note: For trunk channels not in use the service state is 'in-service/idle'.

```
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 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'.

```
status station 11001                                          Page   1 of   3
                              GENERAL STATUS
     Administered Type: 6408D+              Service State: No hardware assigned
        Connected Type: N/A            Parameter Download: pending
             Extension: 11001               SAC Activated? no
                  Port: X                 User Cntrl Restr: none
           Call Parked? no               Group Cntrl Restr: none
      Ring Cut Off Act? no             CF Destination Ext:
Active Coverage Option: 1

          EC500 Status: N/A        Off-PBX Service State: in-service/active
       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 Avaya SES server are shown in the **Signaling** section. In addition, the **Audio** section shows the codec type and the IP addresses of the endpoint directly connected. 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
               Port: T00173            Maintenance Busy? no
 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. 52.160 : 5061

G.711MU     Audio:          192. 45. 53.103  : 6000    192. 45. 53.112 : 6000
            Video:
      Video Codec:
                                             Authentication Type: None
    Audio Connection Type: ip-direct
```

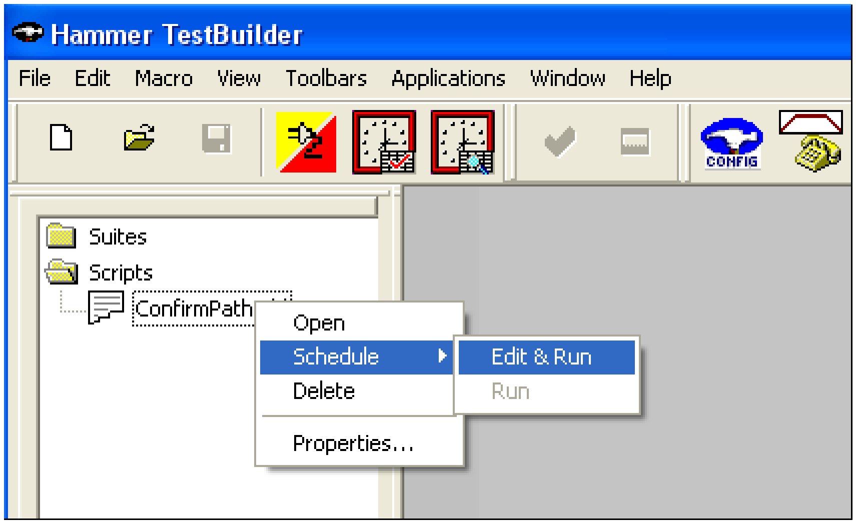
7.2. Verify Avaya SIP Enablement Services

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.

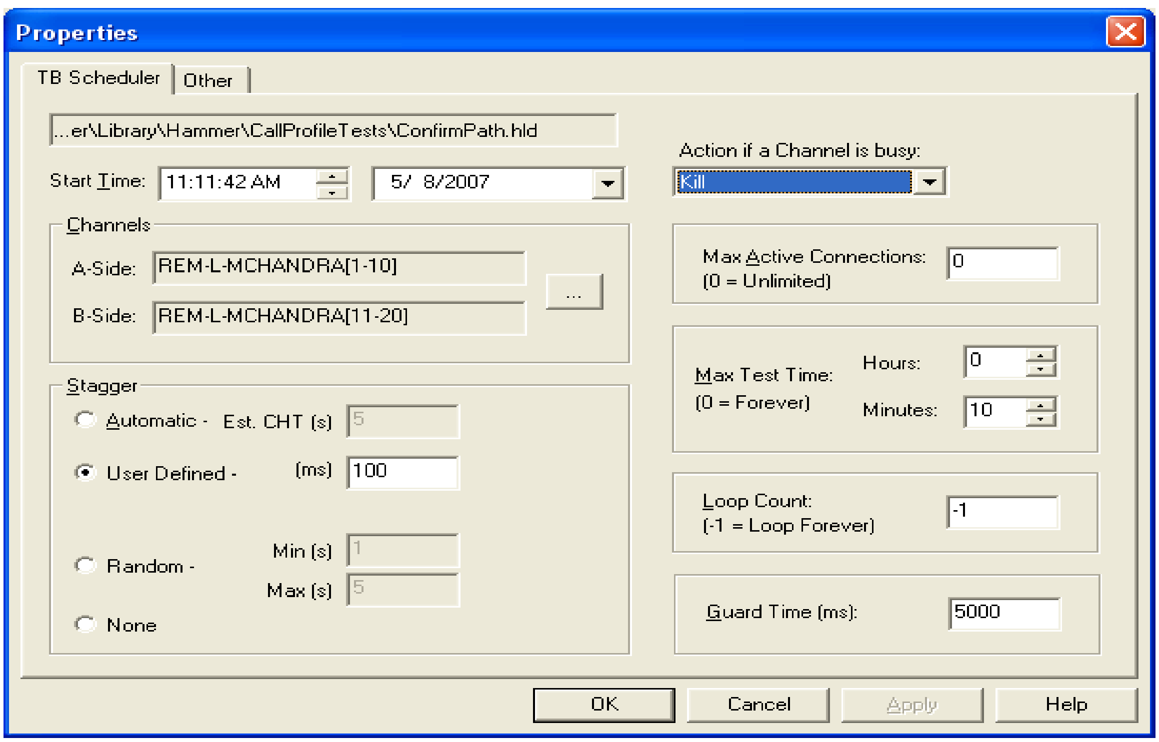
```
craft@CCS-DevCon1> trustedhost -L
Third party trusted hosts.
       Trusted Host       |       CCS Host Name       |         Comment
--------------------------+---------------------------+-------------------------
192.45.53.100             | 192.45.52.160 |
```

7.3. Empirix FX-IP Test Execution

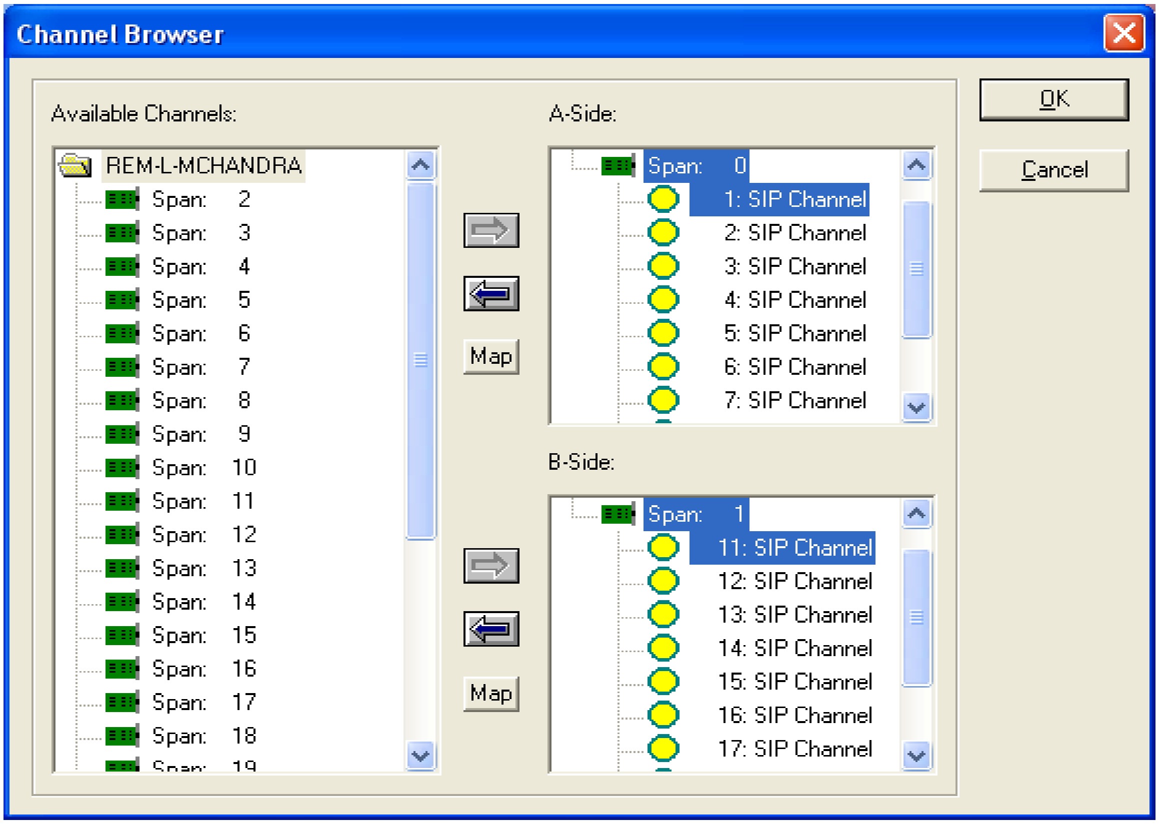
Using the 'Hammer TestBuilder', right click on the 'test case name' from the Right Panel. The 'test case name' will vary. Click on 'Schedule' followed by 'Edit & Run'.



The following screen is displayed. At this point, clicking 'OK' starts the test. The scheduled test should start running.



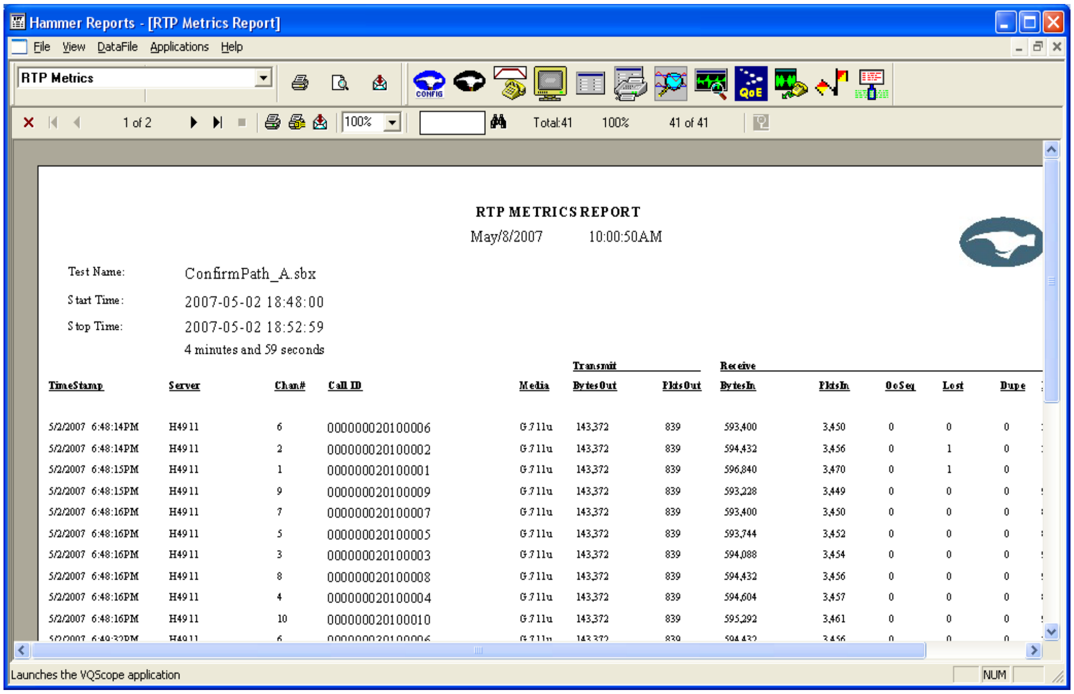
- Select the channels to use to execute the test for 'A-Side' and 'B-Side' as shown in figure below. Click 'OK'.



7.4. Report Generation

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

To generate a report, first select the report type and the report's time frame, then from the drop-down list in the top left-hand corner of the window, select the type of report to view. A dialog box appears that allows selecting parameters for the report, including the time and test data range.



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.

8. Support

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

- Email the Empirix support center via support@empirix.com.
- Call the Empirix support center at 1-800-Empirix.

9. 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.

10. Additional References

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

- *Administrator Guide for Avaya Communication Manager*, Document 03-300509, Issue 1, February 2007, available at http://support.avaya.com

- *Installing and Administering SIP Enablement Services*,, Issue 2.1, March 2007, available at http://support.avaya.com

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Developer*Connection* Program at devconnect@avaya.com.