Avaya Solution & Interoperability Test Lab

Application Notes for Spirent Abacus 5000 with Avaya Communication Manager using the SIP Interface – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Spirent Abacus 5000 Release 3.2 to successfully interoperate with Avaya Communication Manager 3.0.1 using the SIP interface via Avaya SIP Enablement Services 3.0. 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

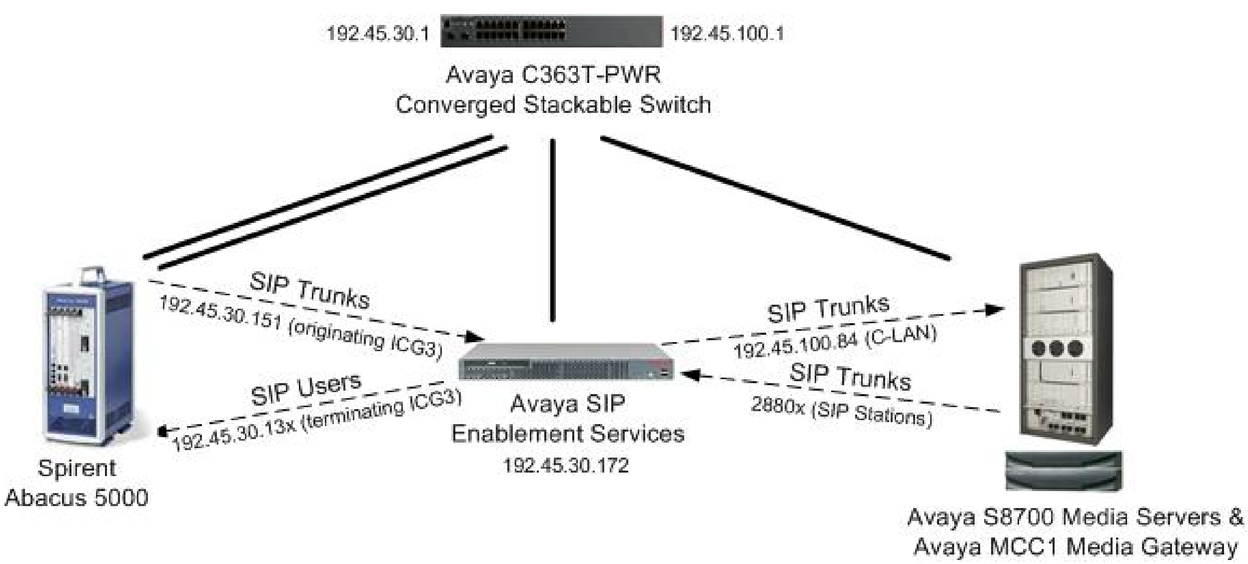
Spirent Abacus 5000 is an integrated IP and PSTN telephony test system with analog, TDM, and Ethernet interfaces. The system generates real voice streams to simulate real-world loads, and performs real time voice quality measurements.

Abacus 5000 can function as a call generator or a switch. The compliance testing focused on Abacus 5000 as a SIP call generator to load Avaya Communication Manager via Avaya SIP Enablement Services (SES). In the SIP integration of Abacus 5000 with Avaya Communication Manager, Abacus 5000 utilizes a SIP capable ICG3 card to originate calls to Avaya SES, and another SIP capable ICG3 card to terminate the calls.

The originating SIP calls from Abacus 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 MCC1 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 Abacus 5000 where the calls are terminated.

Each SIP call in this scenario requires two SIP trunks on Avaya Communication Manager. The ports on the Abacus 5000 terminating ICG3 card are administered as SIP users on Avaya SES, and therefore register with Avaya SES.

For the compliance testing, two separate ICG3 cards are used to originate and terminate the SIP calls on Abacus 5000. With Abacus 5000 Release 3.2 Patch 14, there is software and firmware support for a new ICG3 card with different rear panel that can host two LAN connections. A single such card can be used in place of two separate ICG3 cards.



The compliance testing involved originating calls from the Abacus 5000 across a SIP trunk and terminating these same calls on the Abacus 5000 as SIP endpoints. The information from these Application Notes can easily be extended to other possible test scenarios such as:

Originating From	Terminating To
Abacus Trunks	Avaya SES Users
Abacus Trunks	Abacus Trunks
Abacus Endpoints	Abacus Endpoints

1.1. Abacus 5000 ICG3 Capacity

For the compliance testing, five simultaneous calls were configured and launched on Abacus 5000, thus involving 5 endpoints/channels on each of the originating and terminating ICG3 cards. Below is a table listing of the capacity of the ICG3 card from the Abacus 5000 documentation:

Channel Densities of Abacus 5000 VoIP Subsystems

CG Acronym	CG Subsystem Type	Ethernet Ports per Subsystem	Endpoints (Channels) per Subsystem
ICG3	Voice (PESQ)	1	128
ICG3	Voice (PSQM)	1	256
ICG3/ICL3	RTP (using packet path confirmation only)	1	1,024
ICG3/ICL3	Signaling only	1	4,096 (8,192 for SIP only)

Users need to keep in mind that the endpoint capacity for the ICG3 card can also be impacted by the complexity of the WAV file used for voice confirmation, and by the signaling modes used for communication.

2. Equipment and Software Validated

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

Equipment	Software
Avaya S8700 Media Servers	Communication Manager 3.0.1, load 346.0
Avaya MCC1 Media Gateway • TN799DP C-LAN Circuit Pack • TN2302AP IP Media Processor Circuit Pack	 HW01 FW015 HW13 FW095
Avaya C363T-PWR Converged Stackable Switch	4.3.12
Avaya SIP Enablement Services	3.0, load 31
Spirent Abacus 5000	3.2, patch 14

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
- Administer SIP 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 are 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 Abacus 5000.

```
display system-parameters customer-options                      Page   1 of  11
                                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: 50    0
                    Maximum Off-PBX Telephones -   OPS: 50    11
                    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 Abacus 5000, multiplied by two. This is because 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: 1     0
                    Maximum G250/G350/G700 VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 0     0
          Maximum TN2602 Boards with 320 VoIP Channels: 0     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 Abacus 5000. 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 Abacus 5000 and Avaya Communication Manager from the compliance testing. Retain the default values for the remaining fields on the screen, and submit these changes.

```
change ip-codec-set 7                                           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 Abacus 5000. 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**. 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 MCC1 Media Gateway.

```
change ip-network-region 7                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 7
Location:         Authoritative Domain:
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 7              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 SES Server

Use the "change node-names ip" command, and add entries for the C-LAN and SES server. In this case, "clan-1b04" and "192.45.100.84" are entered as **Name** and **IP Address** for the C-LAN, and "sip-server" and "192.45.30.172" are entered as **Name** and **IP Address** for the 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
aes98               192.45 .95 .98                             .   .   .
cceserver           192.45 .120.15                             .   .   .
clan-1a03           192.45 .100.97                             .   .   .
clan-1b09           192.45 .100.87                             .   .   .
clan-1c04           192.45 .120.140                            .   .   .
clanP2-1a04         192.168.61 .21                             .   .   .
clanP27-2a03        172.16 .252.200                            .   .   .
default             0  .0  .0  .0                              .   .   .
devcon32-1a03       192.45 .100.36                             .   .   .
devcon33-1a03       192.45 .100.16                             .   .   .
ipoffice-room3      192.45 .30 .162                            .   .   .
medpro-1b05         192.45 .100.85                             .   .   .
clan-1b04           192.45 .100.84                             .   .   .
sip-server          192.45 .30 .172                            .   .   .

( 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 1b04" command. Note that the actual slot number may vary. In this case, "1b04" 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 1b04
                                  IP INTERFACES


                  Type: C-LAN
                  Slot: 01B04
           Code/Suffix: TN799  D
             Node Name: clan-1b04
            IP Address: 192.45 .100.84
           Subnet Mask: 255.255.255.0
       Gateway Address: 192.45 .100.1
  Enable Ethernet Port? y
        Network Region: 7
                  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 2001
                               DATA MODULE

  Data Extension: 2001            Name: CLAN 1B04 Data Module
            Type: ethernet
            Port: 01B0417
            Link: 11

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

3.5. Administer SIP Trunk Group

Administer a SIP trunk group to interface with the originating ICG3 card from Abacus 5000. 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 88                                              Page   1 of  20
                                TRUNK GROUP

Group Number: 88                   Group Type: sip          CDR Reports: y
  Group Name: SIP Trunk to SES            COR: 1        TN: 1        TAC: 1088
   Direction: two-way       Outgoing Display? n
 Dial Access? n               Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                                       Signaling Group:
                                                     Number of Members: 0
TRUNK PARAMETERS

     Unicode Name? y

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
```

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. Submit these changes.

- **Group Type:** "sip"
- **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 88                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 88             Group Type: sip
                        Transport Method: tls



   Near-end Node Name: clan-1b04            Far-end Node Name: sip-server
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 7
       Far-end Domain: devconnect.com

                                             Bypass If IP Threshold Exceeded? n

        DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? y
 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.

```
add trunk-group 88                                                Page   1 of  20
                                TRUNK GROUP

Group Number: 88                   Group Type: sip           CDR Reports: y
  Group Name: SIP Trunk to SES            COR: 1        TN: 1        TAC: 1088
   Direction: two-way       Outgoing Display? n
 Dial Access? n               Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                                       Signaling Group: 88
                                                     Number of Members: 10
TRUNK PARAMETERS

     Unicode Name? y

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
```

3.8. Administer SIP Stations

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 28801                                               Page   1 of   4
                                     STATION

Extension: 28801                           Lock Messages? n          BCC: 0
     Type: 6408D+                          Security Code:             TN: 1
     Port: X                             Coverage Path 1:            COR: 1
     Name: SIP28801                      Coverage Path 2:            COS: 1
                                         Hunt-to Station:

STATION OPTIONS
              Loss Group: 2              Personalized Ringing Pattern: 1
             Data Module? n                          Message Lamp Ext: 28801
            Speakerphone: 2-way                   Mute Button Enabled? y
        Display Language: english

                                                     Media Complex Ext:
                                                          IP SoftPhone? n
```

Repeat the "add station n" command to add the desired number of SIP stations, which is the same as the desired number of simultaneous SIP calls. For the compliance testing, five SIP stations were administered as shown below. When possible, use consecutive extension numbers for the SIP stations, for ease of configuring Abacus 5000.

```
list station 28801 count 5

                                  STATIONS

Ext/     Port/    Name/                               Room/         Cv1/ COR/    Cable/
 Type    Hunt-to  Surv GK NN            Move    Data Ext      Cv2  COS     Jack

28801    X        SIP28801                                         1
 6408D+                                 no                         1
28802    X        SIP28802                                         1
 6408D+                                 no                         1
28803    X        SIP28803                                         1
 6408D+                                 no                         1
28804    X        SIP28804                                         1
 6408D+                                 no                         1
28805    X        SIP28805                                         1
 6408D+                                 no                         1
```

For each SIP station created above, have calls to each SIP station be routed to Avaya SES by using the "change off-pbx-telephone station-mapping n" command, where "n" is the station extension. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **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.

```
change off-pbx-telephone station-mapping 28801                    Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station     Application   Dial   Phone Number      Trunk       Configuration
 Extension                 Prefix                   Selection   Set
 28801         OPS              - 28801              88         1
                                -
                                -
```

Repeat the "add off-pbx-telephone station-mapping n" command for all created SIP stations, as listed below.

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

                    STATION TO OFF-PBX TELEPHONE MAPPING

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

28801      OPS    28801             1 /     88          both         all
28802      OPS    28802             1 /     88          both         all
28803      OPS    28803             1 /     88          both         all
28804      OPS    28804             1 /     88          both         all
28805      OPS    28805             1 /     88          both         all
```

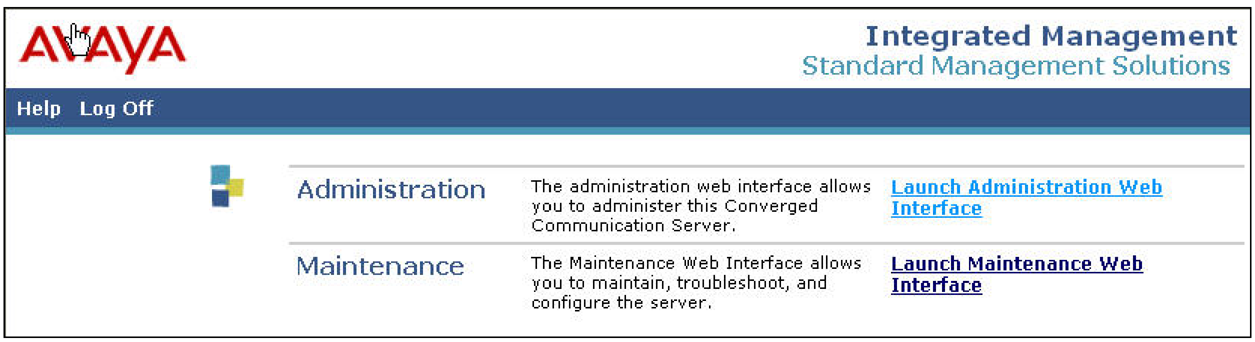
4. Configure Avaya SIP Enablement Services

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

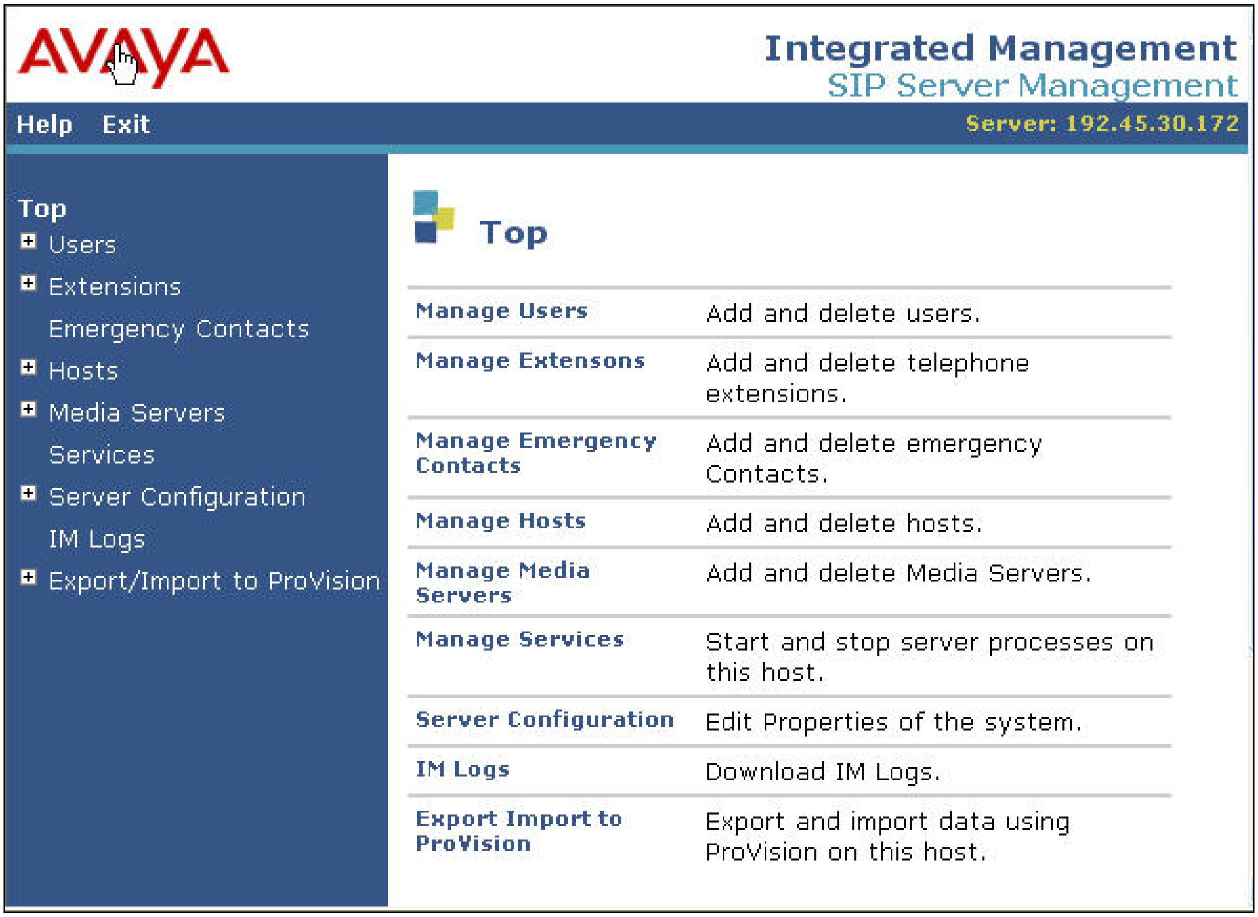
- Obtain SIP domain and host
- Administer media server
- Administer users
- Administer media server address map
- Administer trusted host

4.1. Obtain SIP Domain and Host

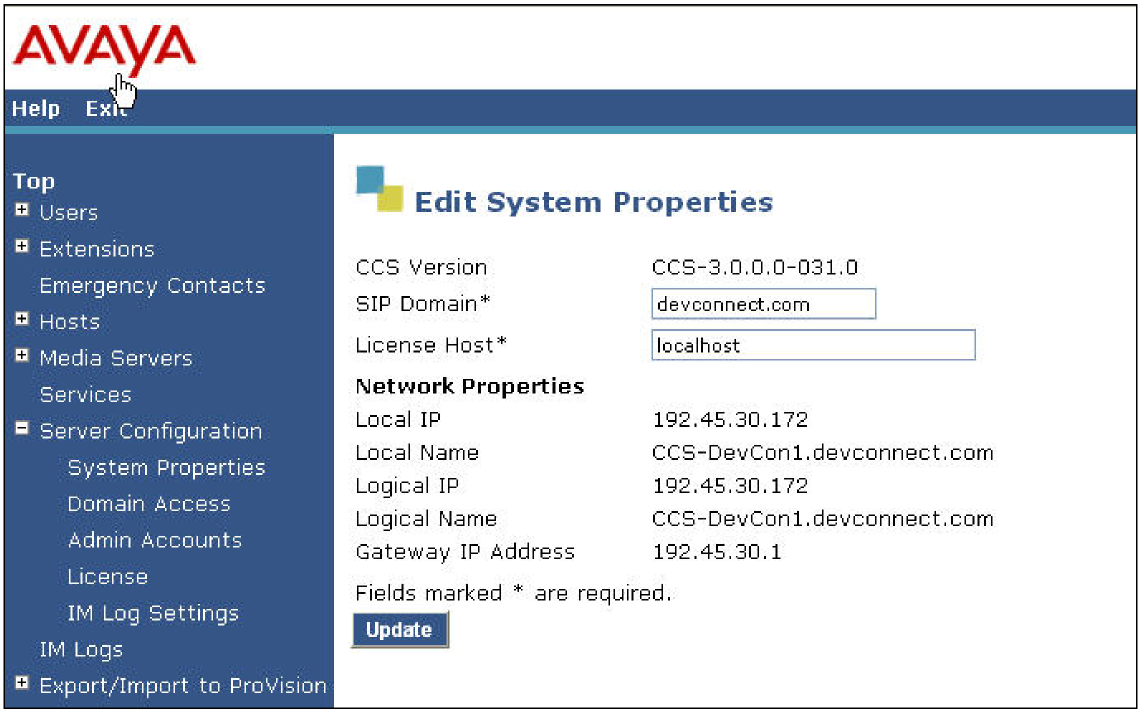
Access the SES administration web interface by using the URL "http://ip-address/admin" in an Internet browser window, where "ip-address" is the IP address of the SES server. Note that the IP address for the SES server may vary, and in this case "192.45.30.172" is used, as administered in **Section 3.3**. Log in with the appropriate credentials and select the **Launch Administration Web Interface** option.



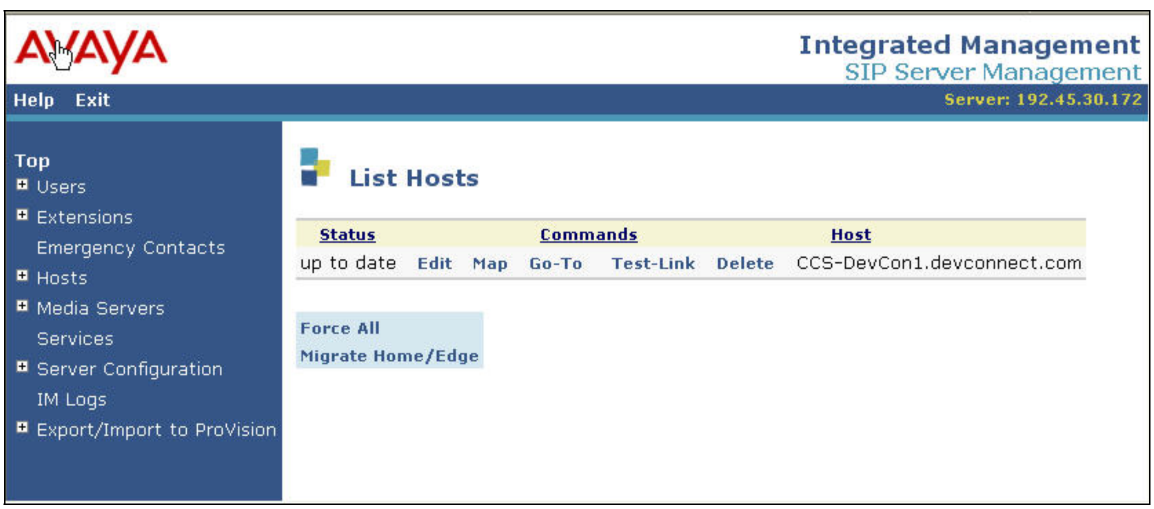
The **Top** screen is displayed next. If this is the initial setup of the SES server, then follow the SIP Enablement Services Installation and Administration Guide [2] to administer the SIP domain and host. These Application Notes assume the SES server has already been configured with the proper domain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



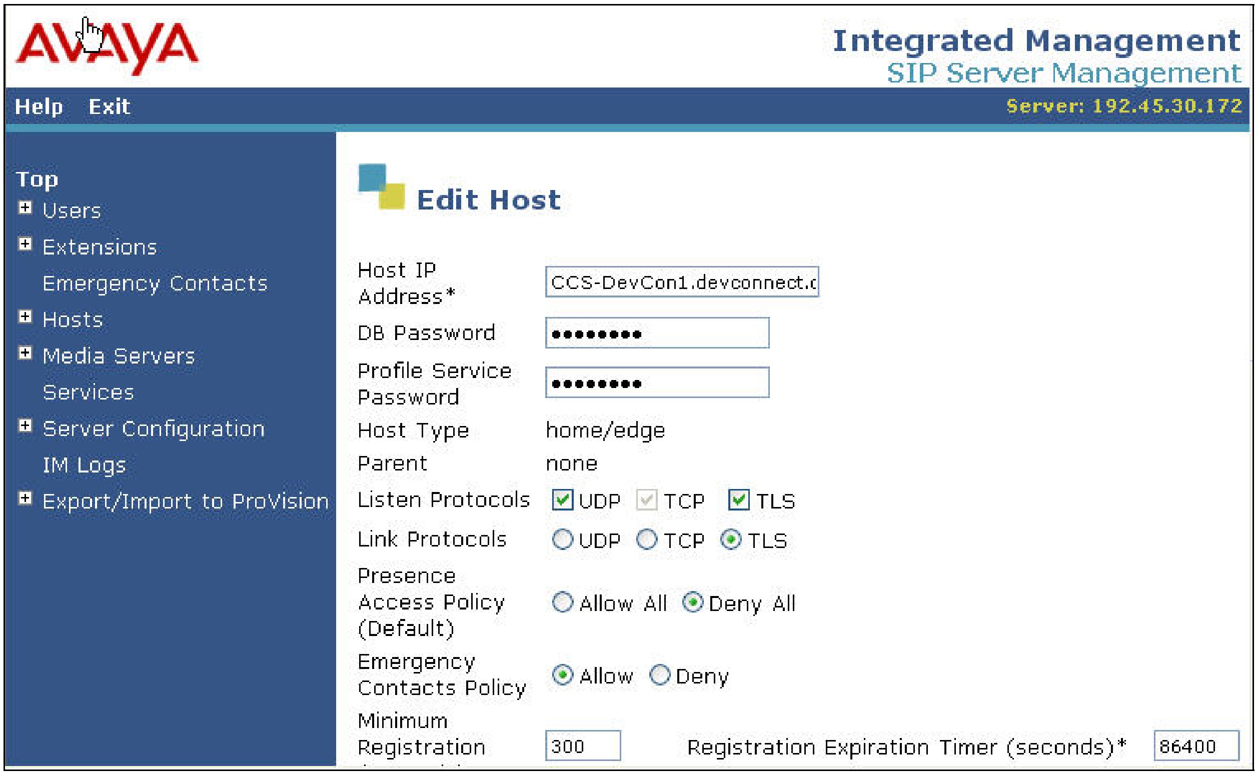
Select **Server Configuration** > **System Properties** from the left pane to display the **Edit System Properties** screen below. Make a note of the value in the **SIP Domain** field, in this case "devconnect.com", as this will be used later to configure Abacus 5000.



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



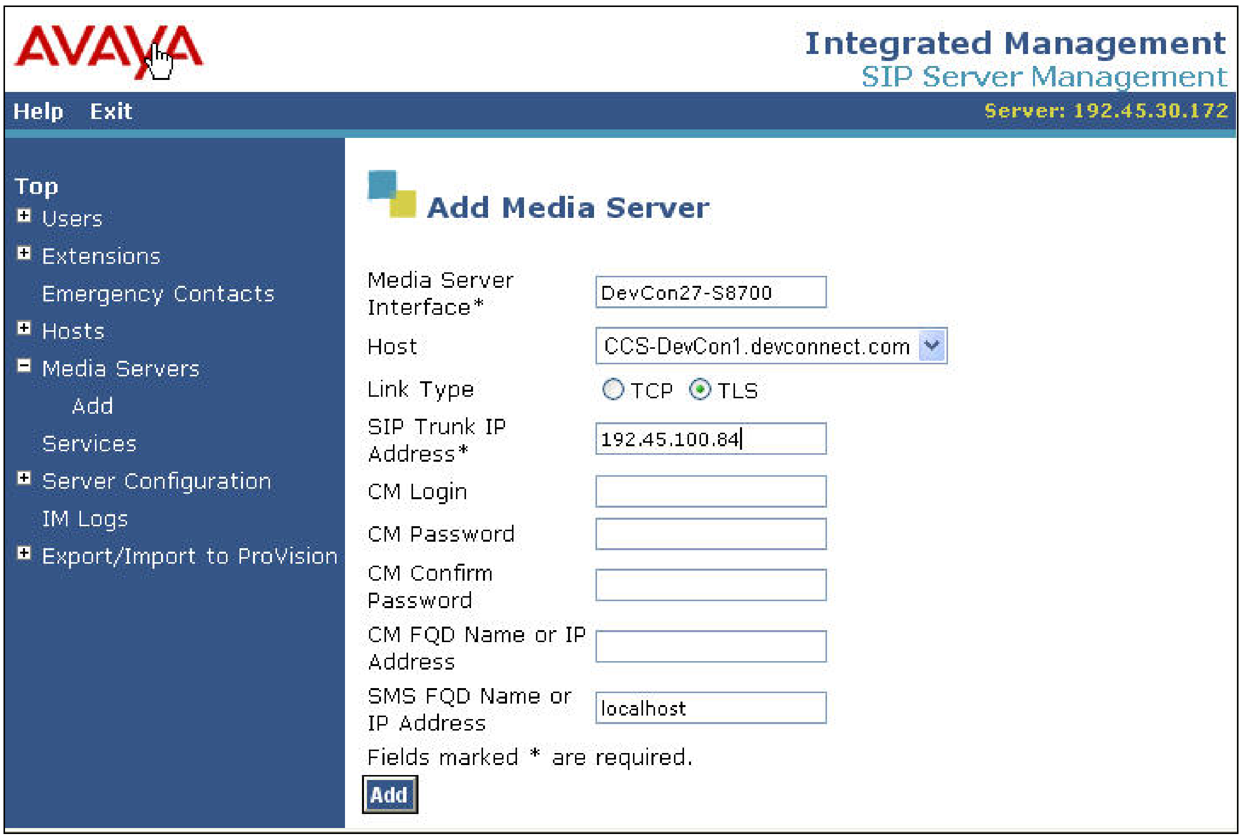
In the **Edit Host** screen, check the value of the **Host Type** field. Perform this check for all hosts listed in the previous **List Hosts** screen, and make a note of the name of the host that is serving as the home server, to be used later to configure the media server interface. For the compliance testing, only one host is administered as both the edge and home server, as indicated by the "home/edge" value in the **Host Type** field shown below. The host name of this home server is "CCS-DevCon1.devconnect.com", indicated in the **Host IP Address** field.



4.2. Administer Media Server

Select **Media Servers** > **Add** from the left pane 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 pane at the end to submit these changes.

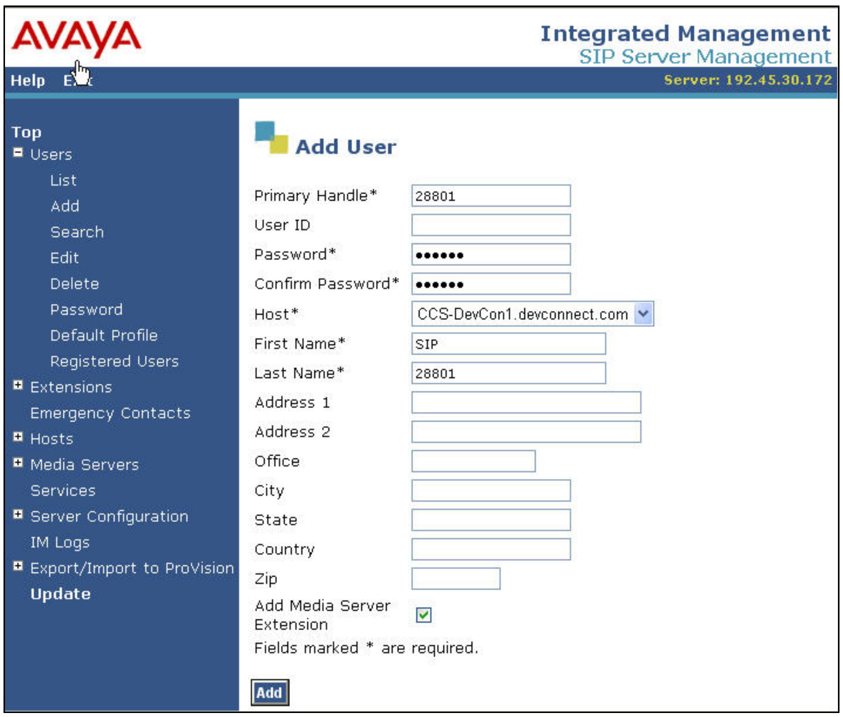
- **Media Server Interface:** A descriptive name.
- **Host:** Select the host name of the home server from **Section 4.1**.
- **SIP Trunk IP Address:** Enter the C-LAN IP address from **Section 3.3**.



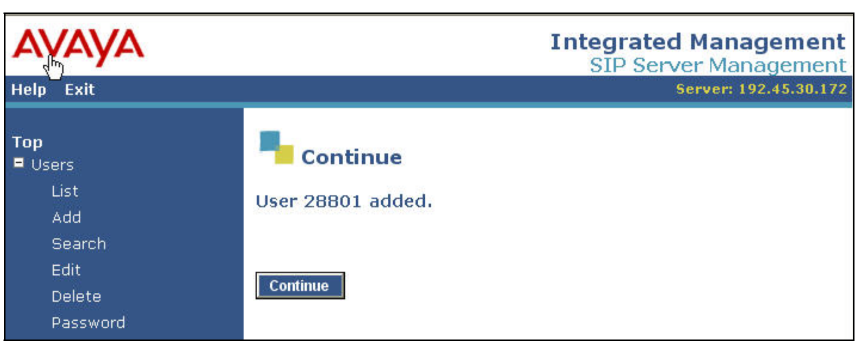
4.3. Administer Users

Select **Users** > **Add** from the left pane to display the **Add User** screen. Enter the following values for the specified fields. 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**.
- **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.

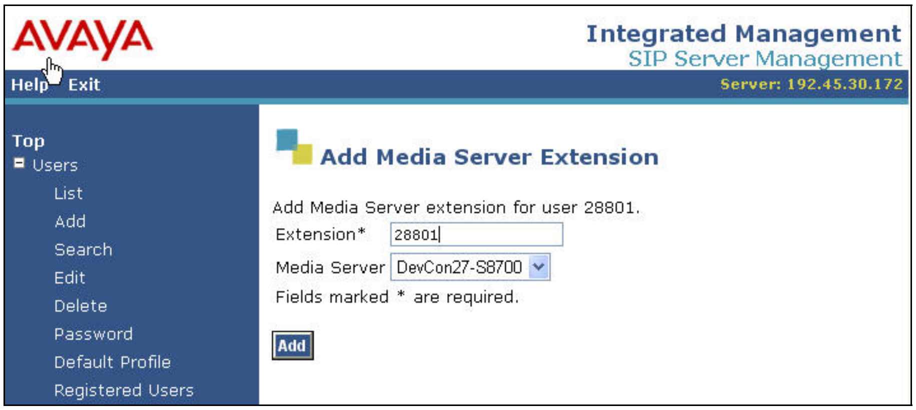


The **Continue** screen is displayed next. Click on the **Continue** button in the bottom of the screen.



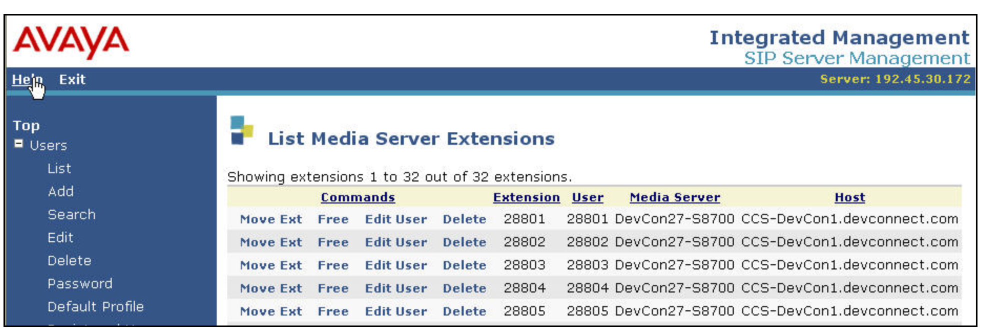
The **Add Media Server Extension** screen is displayed next. This screen is used to associate a user with a media server extension. Enter the following values for the specified fields. Click on **Add** in the bottom of the screen to submit these changes.

- **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. For the compliance testing, five SIP users and extensions were administered as shown below. When possible, use consecutive numbers for the SIP users, for ease of configuring Abacus 5000 later on.



4.4. Administer Media Server Address Map

Select **Media Servers** from the left pane 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 "DevCon27-S8700".

In the **List Media Server Address Map** screen, click on the **Add Map In New Group** link in the lower right pane.

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 28800-28899 are to be routed to media server "DevCon27-S8700". Therefore, the value "DID-288xx" is entered to denote the name of this media server address mapping.

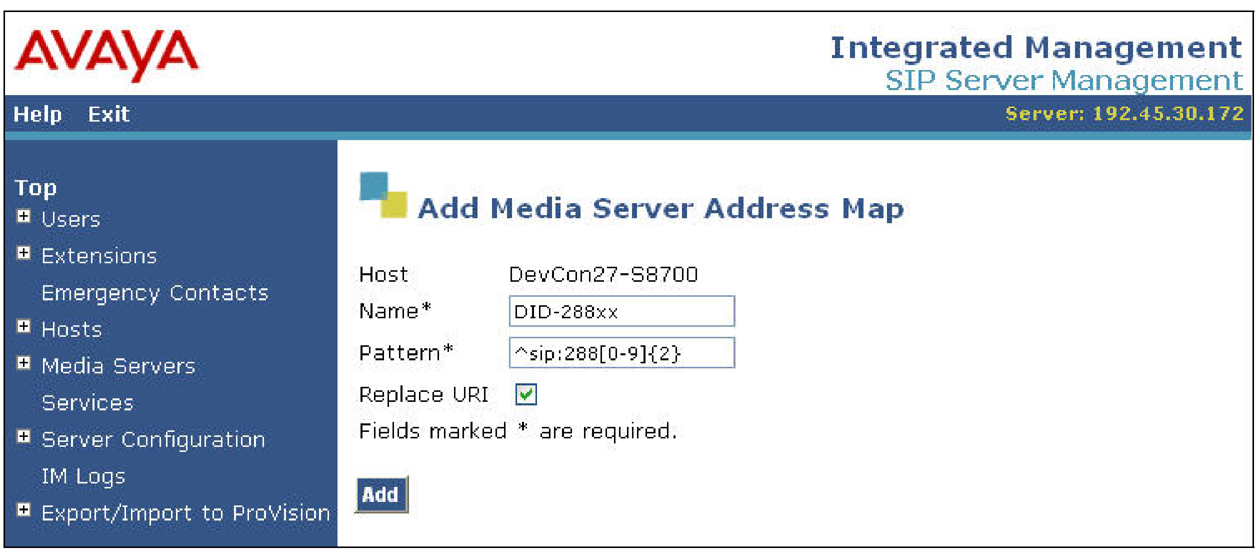
For the **Pattern** field, enter an appropriate syntax for address mapping. The syntax in the **Pattern** field is compared to the Uniform Resource Identifier (URI) of an incoming SIP INVITE message. If a match occurs, then the call is routed to the media server. For the compliance testing, the value "^sip:288[0-9]{2}" was used and explained as follows:

.

- **^sip:** Match to any SIP INVITE message with "sip:" at the beginning of the URI.
- **288** Match to extension digits beginning with "288".
- **{2}** Match to extensions with two more digits following "288".
- **[0-9]** Match the last two digits of the extension to any digits.

Therefore, the pattern "^sip:288[0-9]{2}" will match the extension range of 28800-28899. The actual **Name** and **Pattern** values may vary. The compliance testing could have used "DID-2880x" with "^sip:2880[1-5]{1}" as the **Name** and **Pattern**, to strictly limit extensions 28801-28805 to be routed to the media server. 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 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 Abacus 5000 as a trusted host, so that the SIP Bye messages from Abacus 5000 will not be challenged by 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 originating ICG3 card from **Section 5.2.2**, and “y” is the host name of the SES home server from **Section 4.1**.

```
craft@CCS-DevCon1> trustedhost -a 192.45.30.151 -n CCS-DevCon1.devconnect.com
192.45.30.151 is added to trusted host list.
```

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

5. Configure Abacus 5000

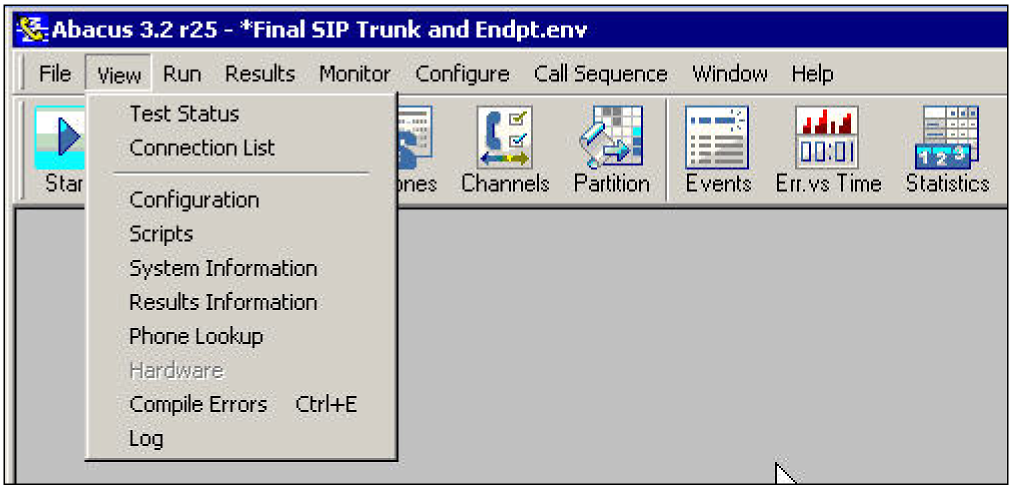
The procedures for configuring the originating and terminating SIP interfaces on Abacus 5000 to interoperate with Avaya Communication Manager fall into the following areas:

- Verify system information
- Administer protocol selection
- Administer phones
- Administer channels
- Administer partitioning and timing

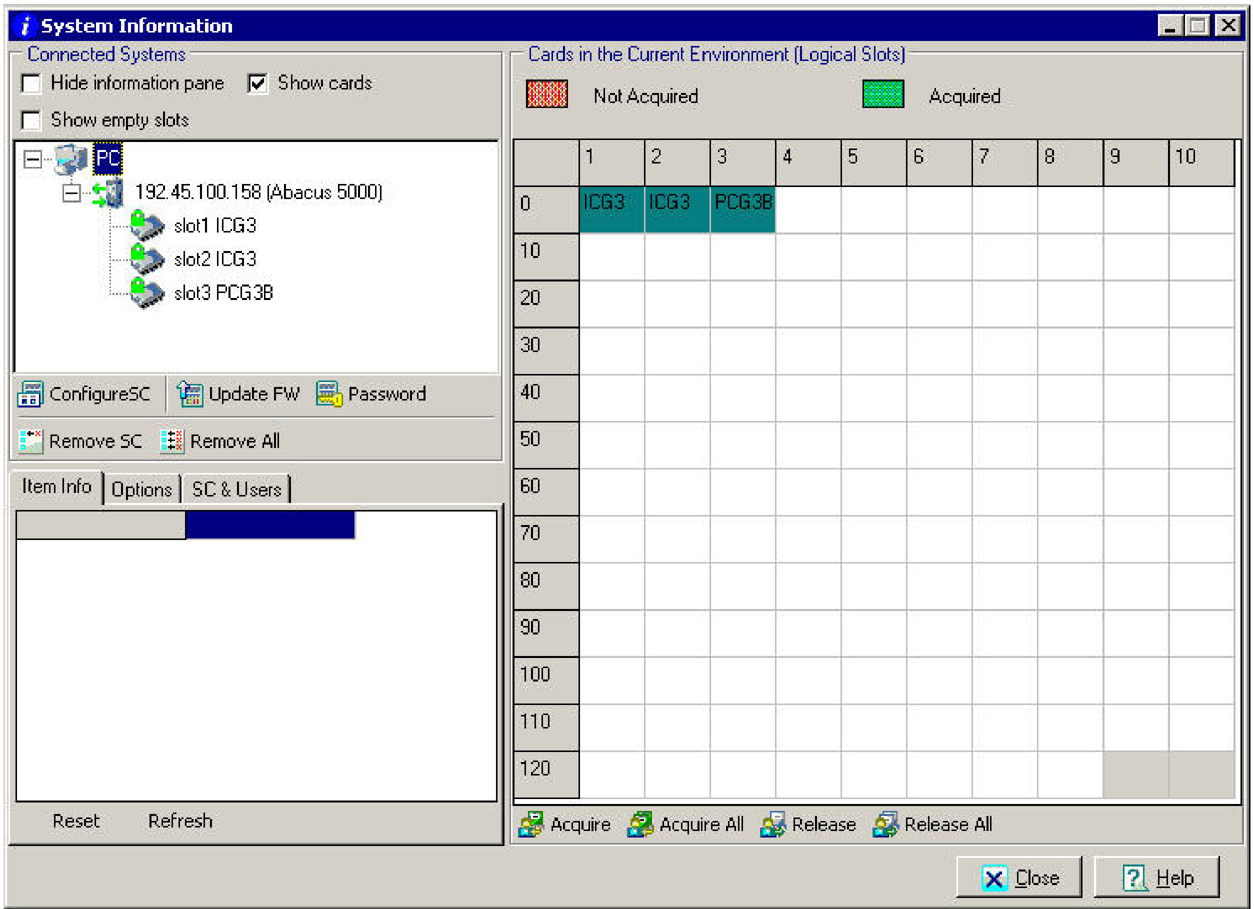
As part of the installation, the Abacus 5000 user interface software is installed on a standalone PC, and used to configure the IP address for the system and to connect to the system.

5.1. Verify System Information

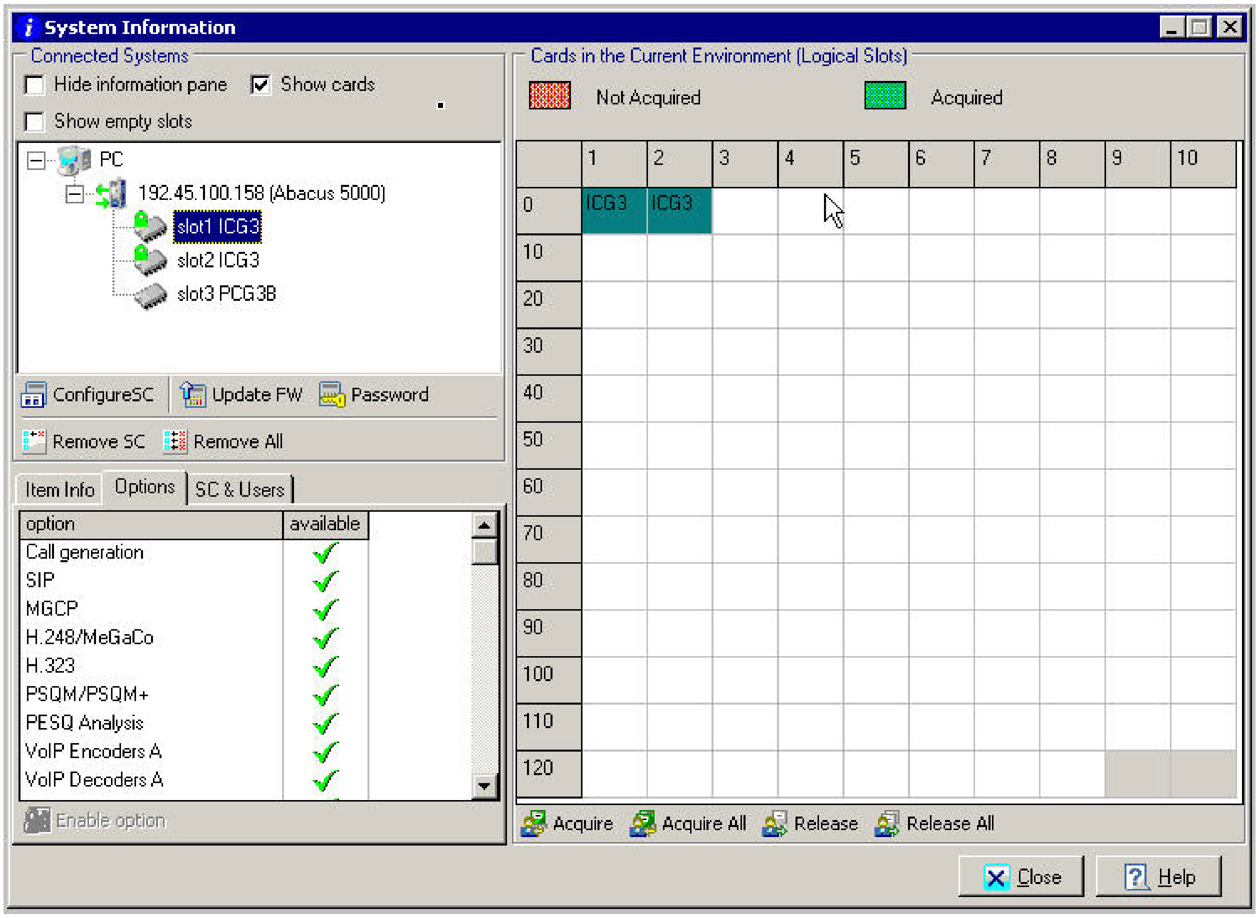
From the PC running the Abacus 5000 user interface, navigate to **Start** > **Programs** > **Abacus 5000** > **3.2** > **Abacus** to open the Abacus 5000 window. Select **View** > **System Information** from the main menu bar as shown below.



The **System Information** screen is displayed, and shows the cards that are available in the system. For the compliance testing, two ICG3 cards are used, one to originate SIP calls over to Avaya SES, and the other to terminate the SIP calls. Release any extra card by selecting the green grid that corresponds to the card in the right pane, in this case "PCG3B", and click on the **Release** button in the bottom of the pane to release the card. Repeat this procedure to release all extra cards.

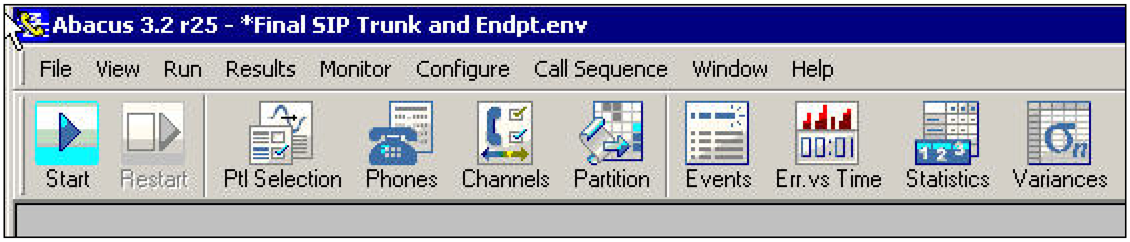


Next, verify the license on each ICG3 card by selecting the **ICG3** card in the directory pane in the upper left section of the window, and clicking on the **Options** tab in the lower left pane to view the available options the card supports. Verify that the **Call generation**, **SIP**, **PSQM/PSQM+** or **PESQ Analysis** options are enabled with a corresponding check mark in the **available** column. Also verify that the appropriate **VoIP Encoders/Decoders** options are enabled, if audio codec other than G.711 is desired (consult the Abacus 5000 documentation for the appropriate codec option). After verifying the options on both of the ICG3 cards, click on **Close**.



5.2. Administer Protocol Selection

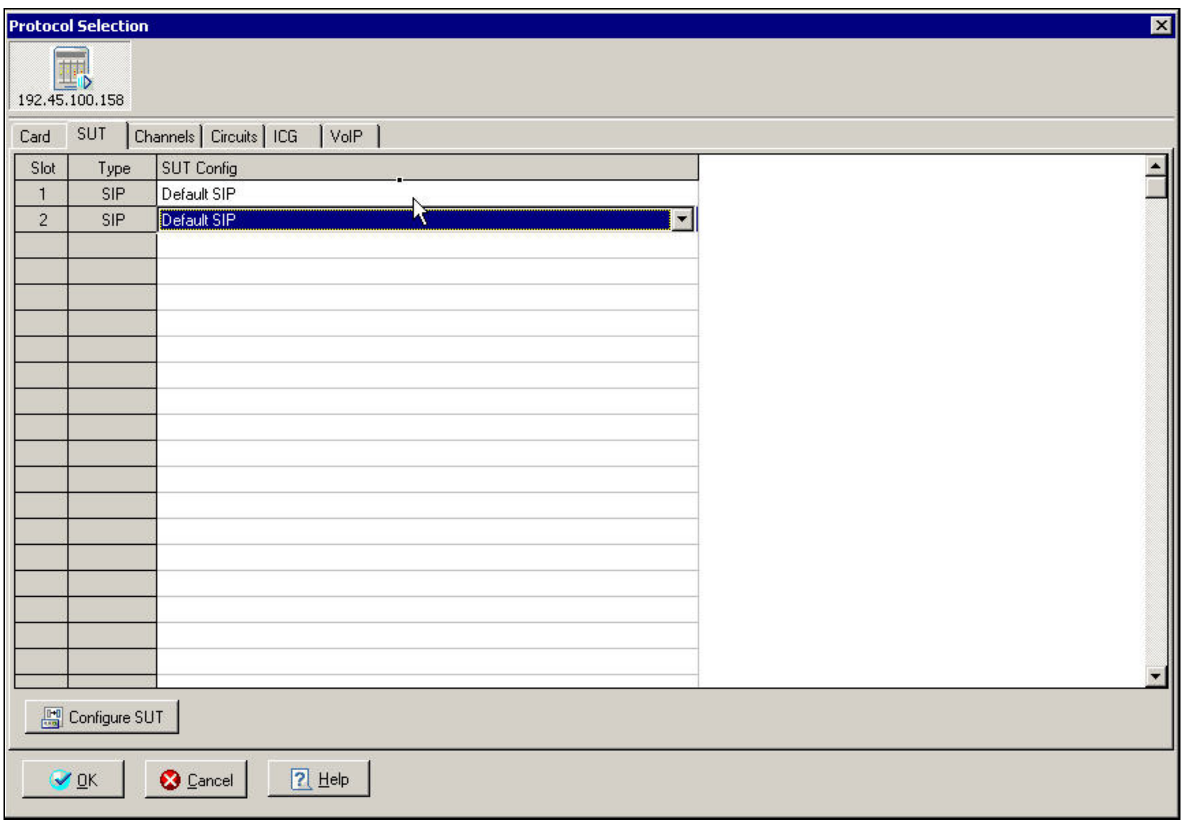
Click on the **Ptl Selection** icon from the main menu bar shown below.



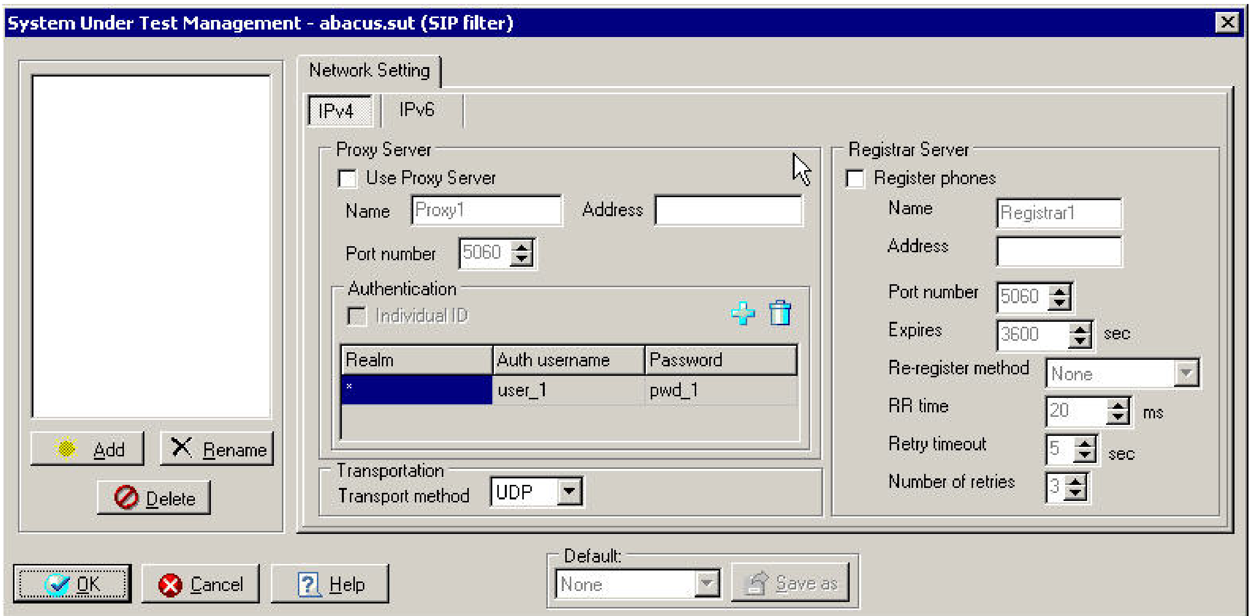
5.2.1. Administer Protocol Selection SUT

Keep the default values in the **Card** tab.

Select the **SUT** tab. Retain the default value in the **SUT Config** field for the first card, to be used to originate the SIP calls. Select the **SUT Config** field for the second card, and click on the **Configure SUT** button in the lower left of the screen.



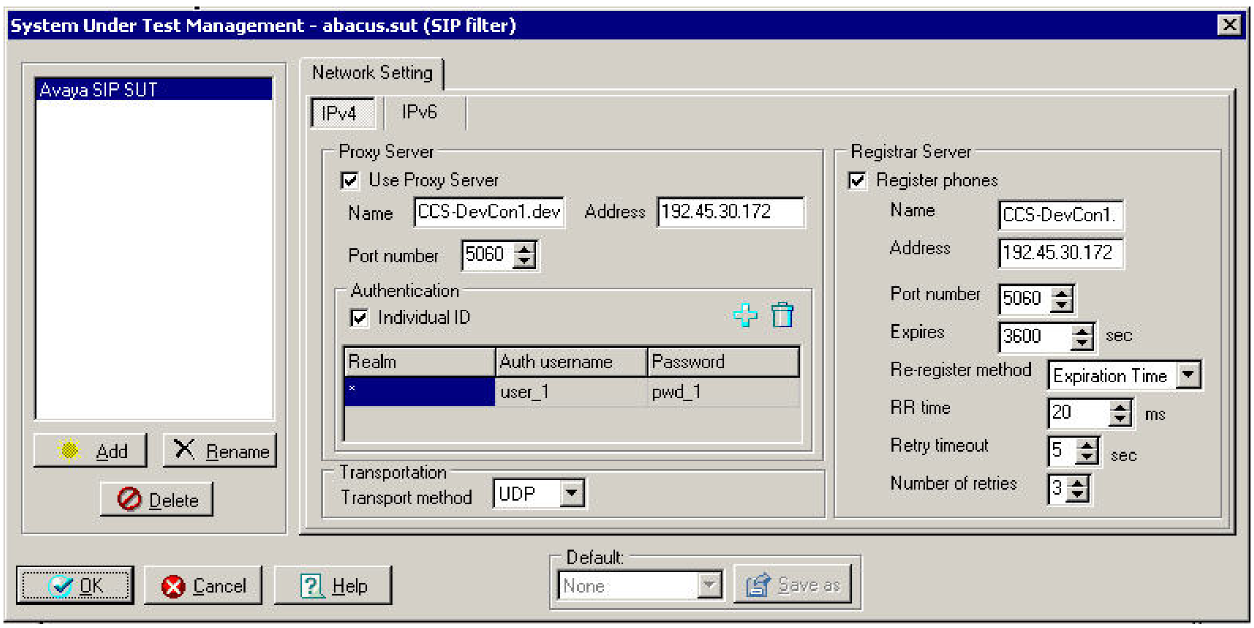
The **System Under Test Management** screen below is displayed on top of the **Protocol Selection** screen. Click on the **Add** button in the lower left pane to add a new SUT for the terminating ICG3 card. This will be used to administer the signaling modes and SES proxy and registrar server information to allow for registration of SIP endpoints on the terminating ICG3 card to SES.



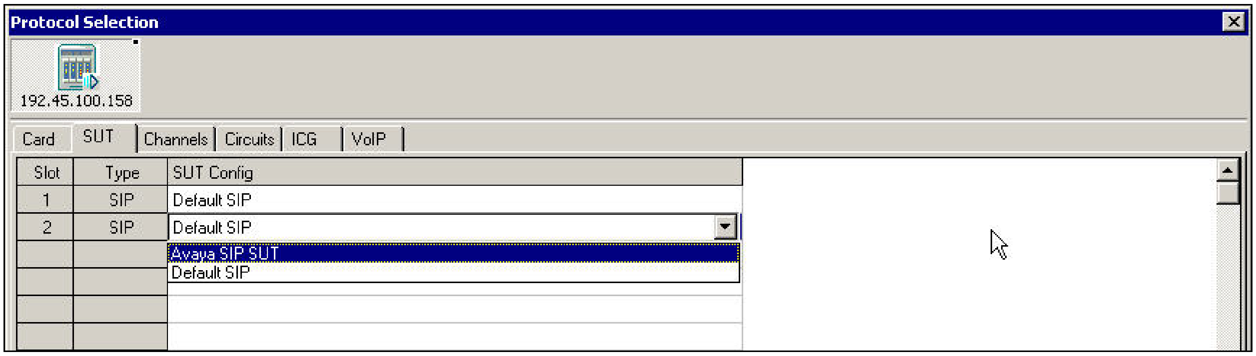
The **New SUT Configuration Record** screen below is displayed on top of the **System Under Test Management** screen. Enter a descriptive name and click on **OK**.

The **System Under Test Management** screen is displayed next, and shows the newly added SUT in the top left pane. Select the **Network Setting** tab, followed by the **IPv4** sub tab. Enter the following values into the specified fields, and retain the default values for all remaining fields. Click on **OK** at the end.

- **Use Proxy Server:** Select this field to enable external proxy server for registration.
- **Name:** Host name of SES home server from **Section 4.1**.
- **Address:** IP address of SES server from **Section 3.3**.
- **Individual ID:** Select this field to enable authentication by individual identifiers.
- **Register phones:** Select this field to enable external register server for registration.
- **Name:** Host name of SES home server from **Section 4.1**.
- **Address:** IP address of SES server from **Section 3.3**.
- **Re-register method:** Select "Expiration Time" from the drop down list.



The **Protocol Selection** screen is displayed next. Click on the **SUT Config** field for the second ICG3 card, and select the newly created SUT from the drop down list. In this case, "Avaya SIP SUT". This enables the terminating ICG3 card to register the SIP endpoints with Avaya SES, using the SES proxy and registrar server information in the newly created SUT.



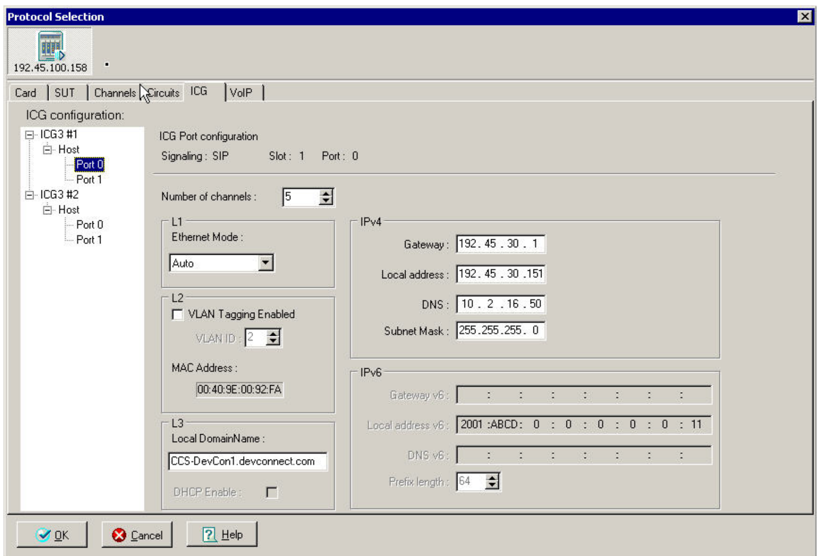
Maintain the defaults in the **Channels** and **Circuits** tabs. Should audio codec other than the default G.711 is desired, then this needs to be administered in the **Channels** tab. The following are the audio codec types that successfully interoperated between Abacus 5000 and Avaya Communication Manager during the compliance testing.

Abacus 5000	Avaya Communication Manager
G.711	G.711MU
G.723	G.723-5.3K, G.723-6.3K
G.729AB	G.729AB, G.729B
G.729B	G.729AB, G.729B

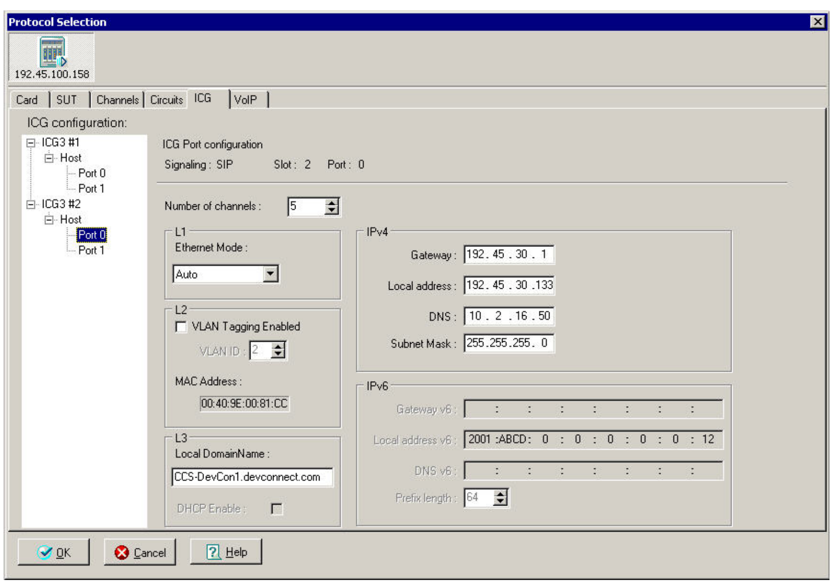
5.2.2. Administer Protocol Selection ICG

Select the **ICG** tab, and click on the **Port 0** field that corresponds to the first ICG3 card. Enter the following values for the specified fields, and retain default values for the remaining fields. Note that the number of channels and IP addresses may vary.

- **Number of channels:** The desired number of originating channels, in this case “5”.
- **Gateway:** Gateway IP address for the network configuration.
- **Local address:** An available IP address, in this case “192.45.30.151”.
- **Subnet Mask:** Subnet mask for the network configuration.

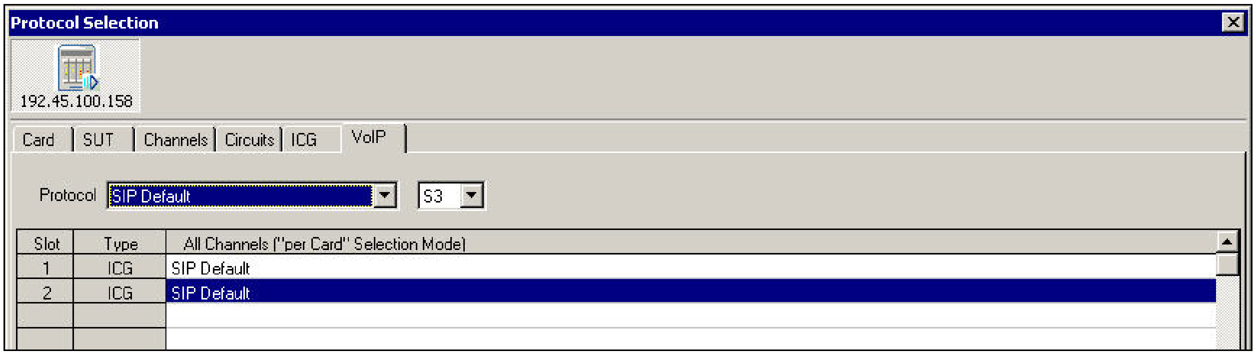


Repeat the same step for **Port 0** in the second ICG3 card, and enter an available IP address for the terminating ICG3 card in the **Local address** field. Click on **OK**.

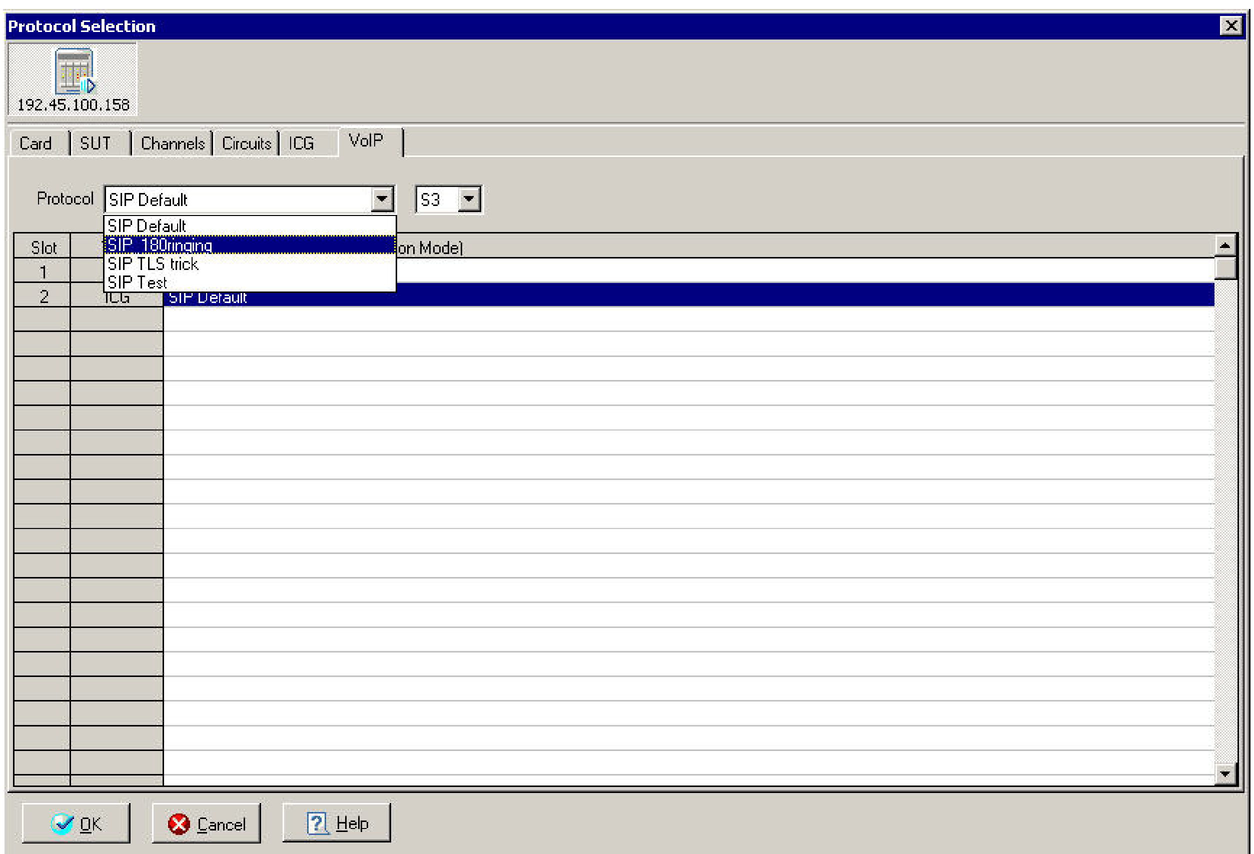


5.2.3. Administer Protocol Selection VoIP

Select the **VoIP** tab. Select the **All Channels (“per Card” Selection Mode)** field for the second ICG3 card, as shown below.

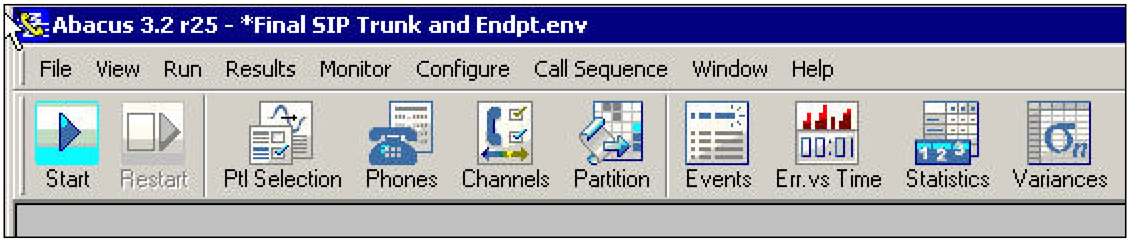


Click on the **Protocol** drop down list to optionally change the default value to “SIP180ringing”. This will enable Abacus 5000 to send a ringing status for incoming SIP calls to the terminating ICG3 card, prior to the actual connection. Click on **OK** to submit the changes. The selection of “SIP 180ringing” for the second ICG3 card is optional, as the default value of “SIP Default” can work as well.



5.3. Administer Phones

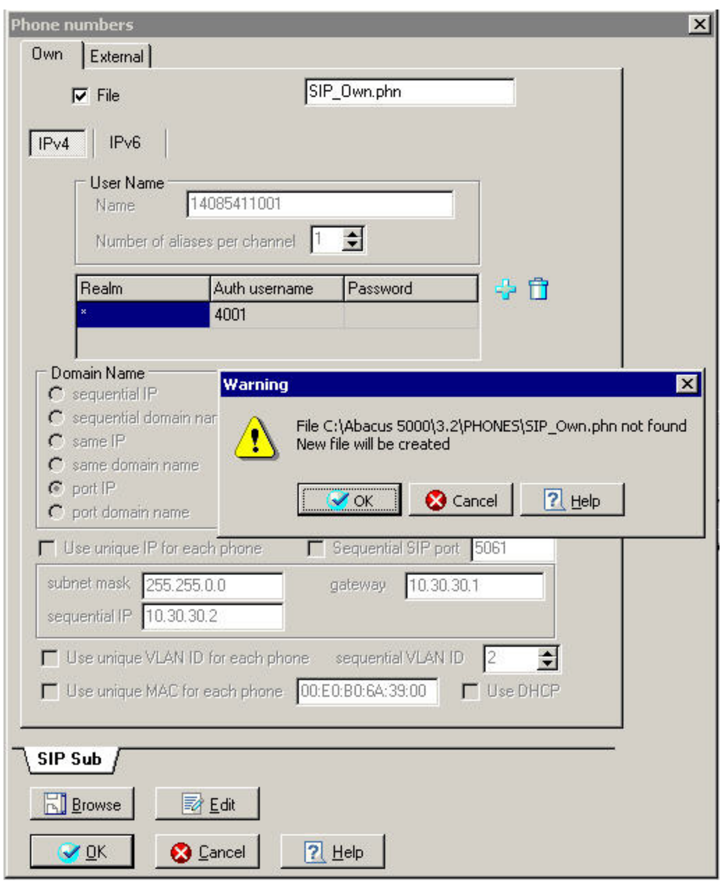
Click on the **Phones** icon from the main menu bar.



5.3.1. Administer Phones Own

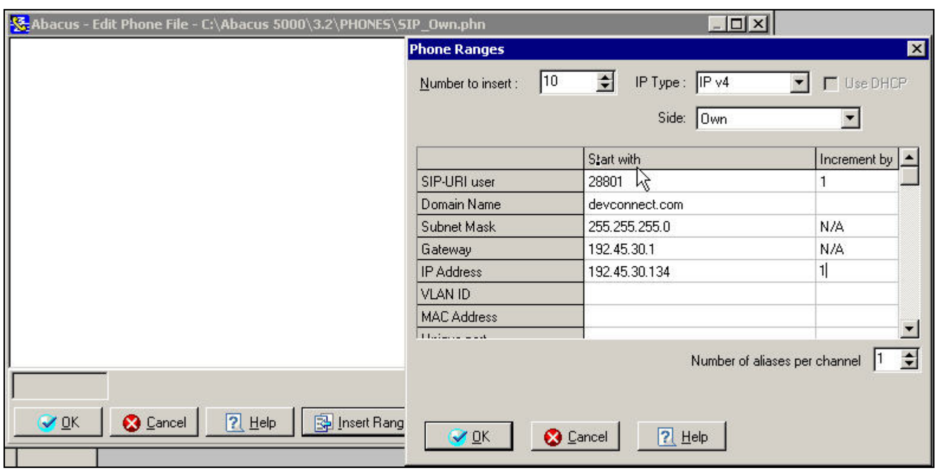
The **Phone numbers** screen is displayed, as shown below. The **Own** tab is used to create internal telephone numbers for the channels on the two ICG3 cards, and the **External** tab is used to create external telephone numbers for the originating channels to dial.

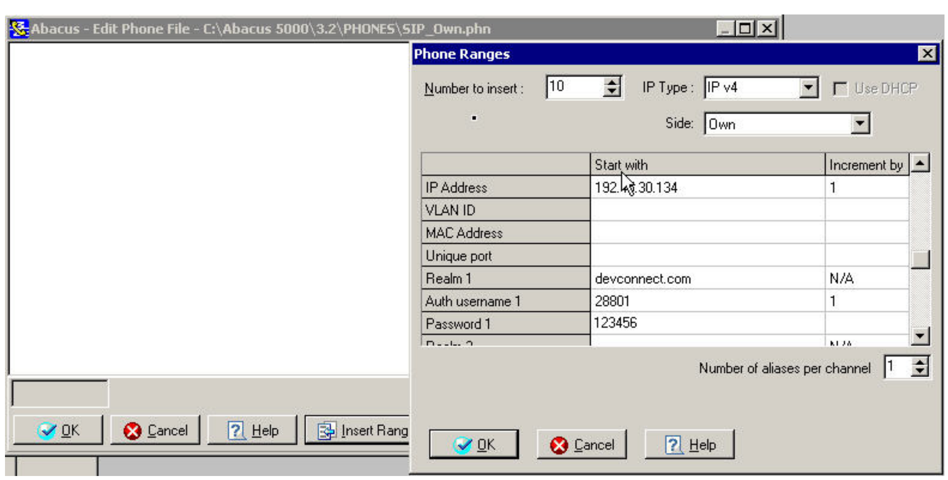
Select the **Own** tab. Click on **File** and replace the default "SIP_Sample.phn" with a desired file name. In this case, "SIP_Own.phn" is used. Click on the **Edit** button at the bottom left of the screen. A **Warning** pop up window is displayed as shown below. Click on **OK** to proceed to create the new file.



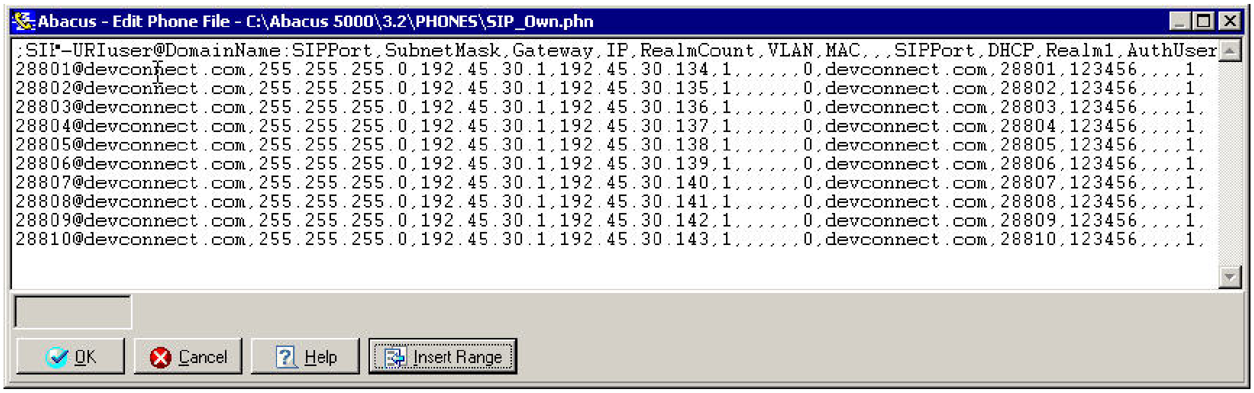
An empty **Edit Phone File** screen is displayed next. Click on **Insert Range** at the bottom of the screen to display the **Phone Ranges** screen as shown below. Scroll down the **Phone Ranges** screen as needed to enter the following values, and click on **OK** at the end.

- **Number to insert:** The total number of originating and terminating channels.
- **IP Type:** Select "IP v4" from the drop down list.
- **Side:** Select "Own" from the drop down list.
- **SIP-URI user:** The starting SIP station number from **Section 3.8**.
- **Increment by:** Incremental value for the station extensions.
- **Domain Name:** The SIP domain name from **Section 4.1**.
- **Subnet Mask:** Subnet Mask for the network configuration.
- **Gateway:** Gateway for the network configuration.
- **IP Address:** The starting value of a series of available IP addresses.
- **Increment by:** Incremental value for the available IP addresses.
- **Realm 1:** The same value as the **Domain Name** field.
- **Auth username1:** The same value as the **SIP-URI user** field.
- **Increment by:** The same value as the **Increment by** field for **SIP-URI user**.
- **Password:** The SIP user password from **Section 4.3**.



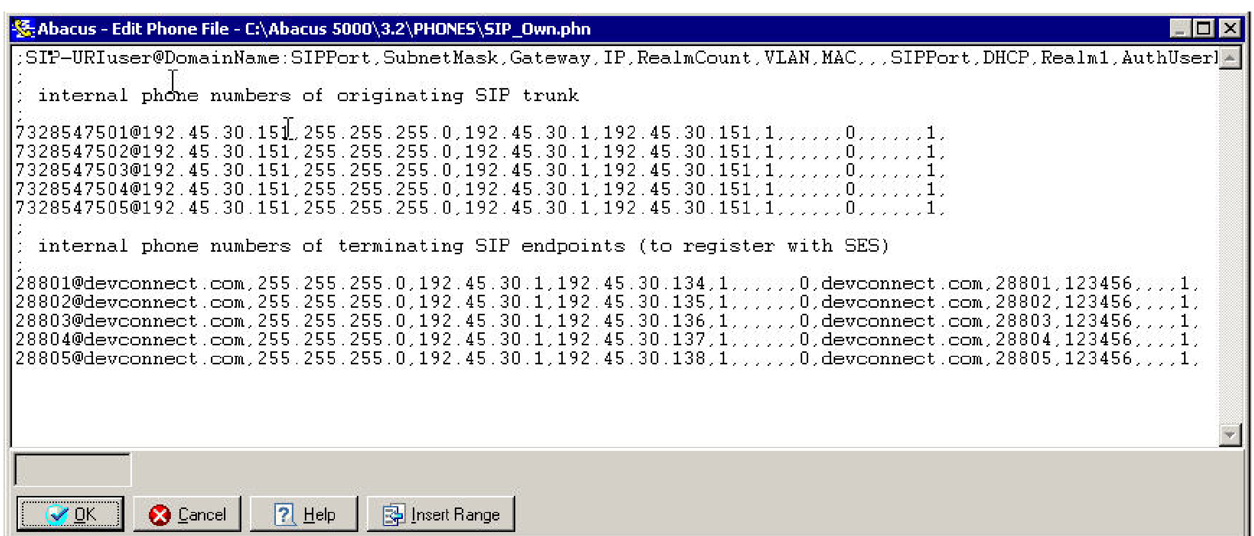


The **Edit Phone File** screen is displayed and filled in with the information entered from the **Phone Ranges** screen. Manually edit the file as necessary.



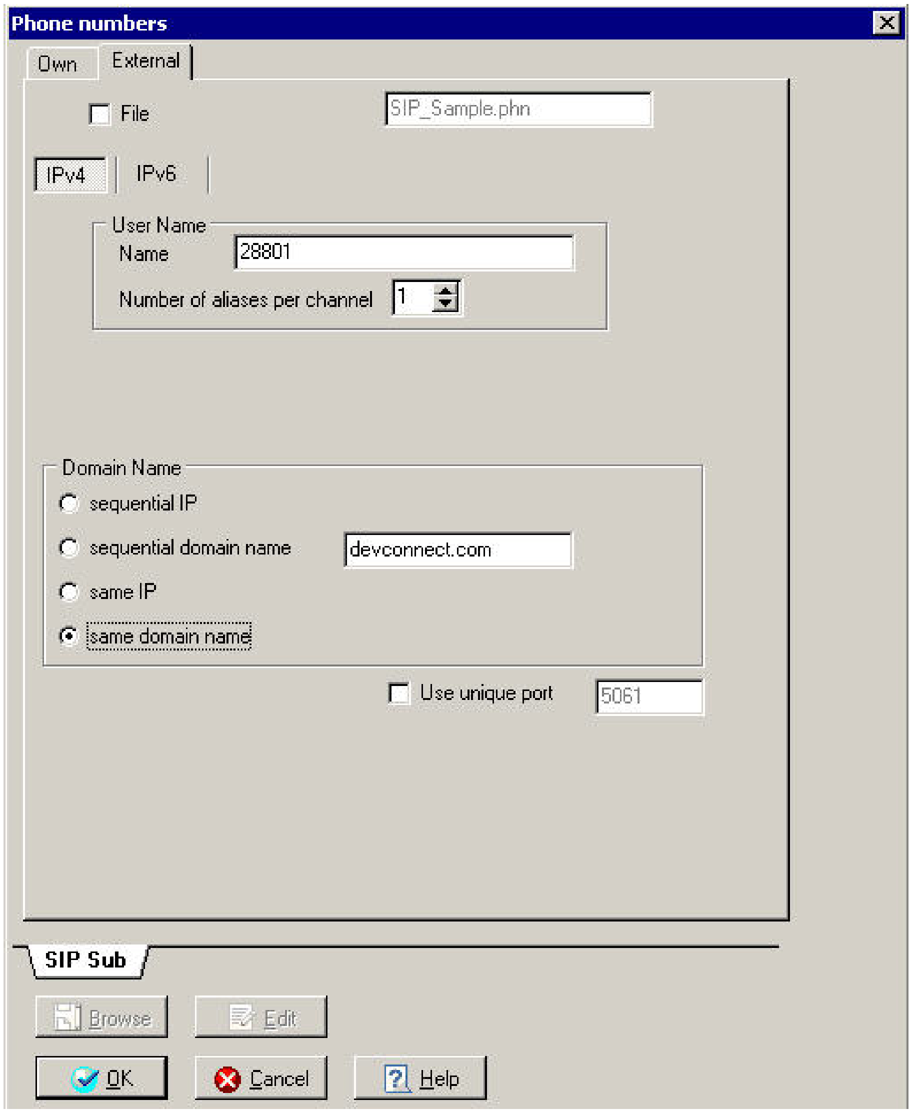
Below is the result of the file after it has been manually edited. Note that lines preceded by ";" are the comment lines, and as many comment lines as desired may be added. The first five entries are the internal telephone numbers for the five channels on the originating ICG3 card. The "288xx" numbers are replaced with the desired digits to be passed to Avaya Communication Manager as calling party numbers. The "devconnect.com" strings and the **IP** values are replaced by the IP address of the originating ICG3 card from **Section 5.2.2**. Remove the values for the **Realm1**, **AuthUserName1** and **Password1** fields (not all field labels are visible in the screen below), as no external registration will be necessary.

The next five entries are the internal telephone numbers for the five channels on the terminating ICG3 card. The "288xx" extensions and **AuthUserName1** should match the SIP user handle created in **Section 4.3**. The **Password1** should match the SIP user password also administered in **Section 4.3**. The **IP** are changed to unique and available IP addresses in the network configuration. Click on **OK**.



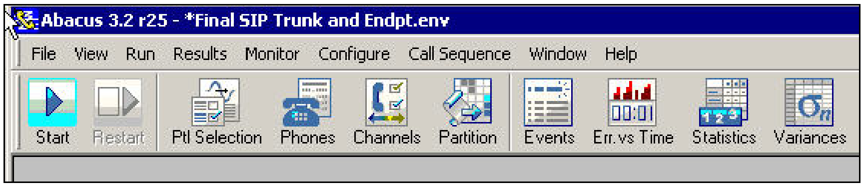
5.3.2. Administer Phones External

Select the **External** tab. For the **Name** field under the **User Name** section, enter the first sequential number of the SIP stations on Avaya Communication Manager from **Section 3.8**. In the case that the SIP station numbers are not sequential, then a file needs to be created and manually edited, similar to the steps that were taken in the previous **Section 5.3.1**. For the **Domain Name** section, select **same domain name** and enter the SIP domain name from **Section 4.1**. In this case, "devconnect.com" is entered as shown below. The values specified for the **Name** and **Domain Name** fields will be used to construct the URI in the SIP message. Maintain the default values for the remaining fields, and click on the **OK** button to submit these changes.



5.4. Administer Channels

Click on the **Channels** icon from the main menu bar.

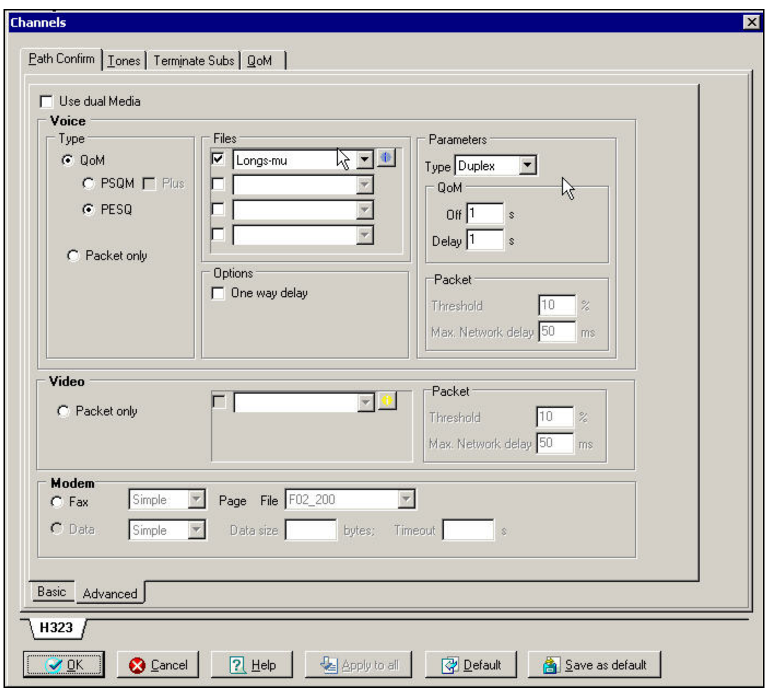


5.4.1. Administer Channels Path Confirm

The **Channels** screen is displayed next. The **Path Confirm** tab is used to select the two-way speech path verification mechanism for the originating and terminating channels for each call. The remaining tabs are not used and can retain the default values.

Select the **Path Confirm** tab, and click on the **Advanced** sub tab in the bottom left of the screen. Select **QoM**, **PESQ**, and "Duplex" from the **Type** drop down list under **Parameters**. Retain the default values for all remaining fields, and click **OK**.

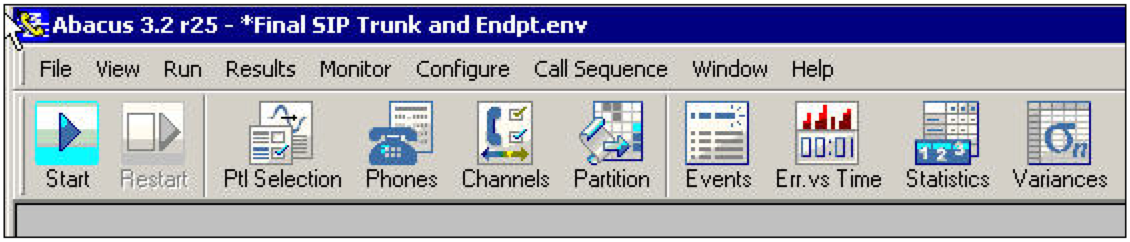
Note that PSQM and PESQ are ITU standards P.861 and P.862 respectively for assessment of speech quality. PESQ is the more recent standard with a higher accuracy than PSQM, based on the comparison results published by the Audio Engineer Society Convention. If PSQM is optioned on the ICG3 card instead of PESQ, then select **PSQM** and enable the **Plus** field. The "Longs-mu" file is a longer WAV file to use for voice path confirmation, and other WAV files may be used. The "Duplex" type is a more stringent test with simultaneous two-way speech on the voice path as opposed to a polite conversation with "Simplex".



 Solution & Interoperability Test Lab Application Notes

5.5. Administer Partitioning and Timing

Click on the **Partition** icon from the main menu bar.



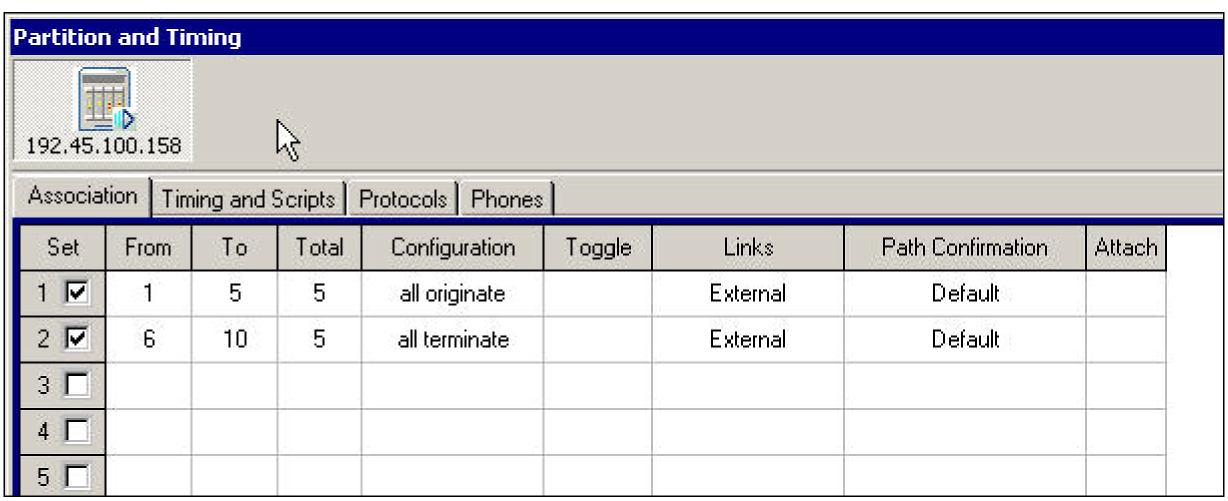
5.5.1. Administer Association

The **Partition and Timing** screen is displayed. Select the **Association** tab, and update the following field:

- **Set:** Click to enter a check mark next to **1** and **2**.
- **From** and **To:** Update to reflect the range of channels on each ICG3 card.
- **Total:** Will be updated automatically by Abacus 5000.
- **Configuration:** Select "all originate" for Set 1, and "all terminate" for Set 2.
- **Links:** Select "External" for both Set 1 and Set 2.

Set 1 corresponds to the originating ICG3 card, which has 5 channels administered in this case (from **Section 5.2.2**), and will be used to originate the SIP calls. Set 2 corresponds to the terminating ICG3 card, which also has 5 channels in this case, and will be used to terminate the SIP calls.

The "Default" value can be retained for the **Path Confirmation** fields, as the path confirmation method has already been administered in **Section 5.4.1**.



5.5.2. Administer Timing and Scripts

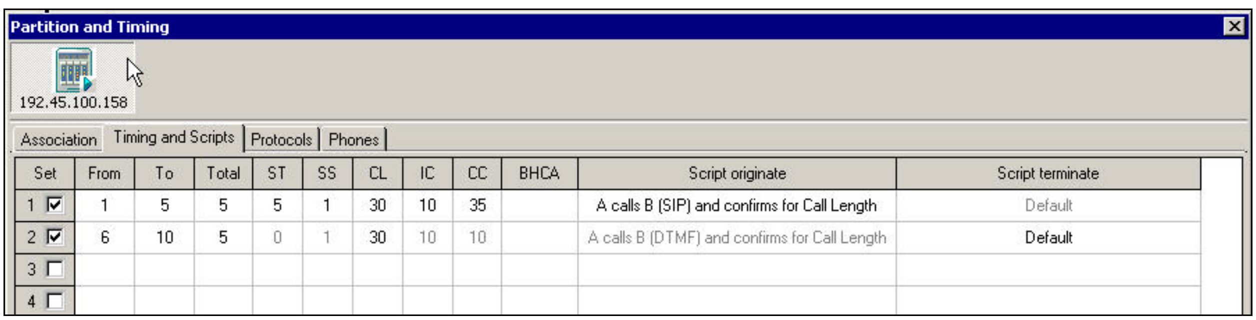
Select the **Timing and Scripts** tab, and update the **ST**, **SS**, **CL**, **IC**, and **CC** fields to the desired values. Below is a brief description of what each abbreviated field stands for from the Abacus 5000 documentation:

		Resolution	Maximum Time
ST	Start	1 second	1023 seconds
SS	Start to Start	0.1 second	99.9 seconds
CL	Call Length	1 second	99999 seconds
IC	Inter-Call	0.1 second	99.9 seconds
CC	Call to Call	1 second	1023 seconds

The value for the **IC** (Inter-Call) field needs to be a minimum of "10" seconds, as this is the amount of time necessary to allow for proper tear down of SIP calls.

Select the script "A calls B (SIP) and confirms for Call Length" for the **Script originate** field for Set 1, and maintain the "Default" value for the **Script terminate** field for Set 2.

Maintain the default values in the **Protocols** and **Phones** tabs, and click on **OK** to submit the changes.



Set	From	To	Total	ST	SS	CL	IC	CC	BHCA	Script originate	Script terminate
1 ☑	1	5	5	5	1	30	10	35		A calls B (SIP) and confirms for Call Length	Default
2 ☑	6	10	5	0	1	30	10	10		A calls B (DTMF) and confirms for Call Length	Default
3 ☐											
4 ☐											

6. Interoperability Compliance Testing

The Interoperability compliance testing focused on the following areas in Abacus 5000:

- Registration of SIP endpoints with Avaya SES.
- 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 Abacus 5000 from Avaya SES.
- Support of various SIP audio codecs with Avaya Communication Manager and SES.
- Voice quality as measured by PESQ scores with path confirmation.
- 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 Abacus 5000 to originate and terminate SIP calls to Avaya Communication Manager via Avaya SES. The audio codec test calls were held up for 90 seconds. The serviceability test cases were performed by disconnecting and reconnecting the LAN cables to the Abacus 5000 originating and terminating ICG3 cards.

The verification included monitoring the various reports from Abacus 5000 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.

There were two observations from the compliance testing. The first is any customized setting of the Protocol Selection SUT will not be preserved in the environment file. The workaround is to manually change the “SIP Default” value corresponding to the second ICG3 card back to the custom SUT, upon each loading of the environment file.

The second observation is that during a test run, when the LAN cable is pulled from the ICG3 card for longer than 30 seconds and then restored, no further calls can be completed. The workaround is to manually stop and restart the test run.

7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of SIP between Avaya Communication Manager and Abacus 5000.

7.1. 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 88

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0088/001 T00534   in-service/active  no   T00073
0088/002 T00535   in-service/active  no   T00179
0088/003 T00538   in-service/active  no   T00181
0088/004 T00539   in-service/active  no   T00074
0088/005 T00073   in-service/active  no   T00534
0088/006 T00074   in-service/active  no   T00539
0088/007 T00179   in-service/active  no   T00535
0088/008 T00180   in-service/active  no   T00182
0088/009 T00181   in-service/active  no   T00538
0088/010 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 88
                         STATUS SIGNALING GROUP

        Group ID: 88                            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”.

```
status station 28801                                           Page   1 of   3
                              GENERAL STATUS
     Administered Type: 6408D+             Service State: No hardware assigned
        Connected Type: N/A           Parameter Download: pending
             Extension: 28801              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 SES server are shown in the **Signaling** section. In addition, the **Audio** section shows the codec type and the IP addresses of the originating ICG3 card and terminating ICG3 channel. The **Audio Connection Type** displays “ip-direct”, indicating media shuffling.

```
status trunk 88/1                                              Page   1 of   2
                              TRUNK STATUS

 Trunk Group/Member: 0088/001              Service State: in-service/active
              Port: T00534              Maintenance Busy? no
 Signaling Group ID:



    Connected Ports: T00538




                   Port     Near-end IP Addr : Port    Far-end IP Addr : Port
        Signaling: 01B0417  192. 45.100. 84  : 5061    192. 45. 30.172 : 5061

G.711MU     Audio:          192. 45. 30.134  : 6000    192. 45. 30.151 : 6000
            Video:
      Video Codec:
                                            Authentication Type: None
    Audio Connection Type: ip-direct
```

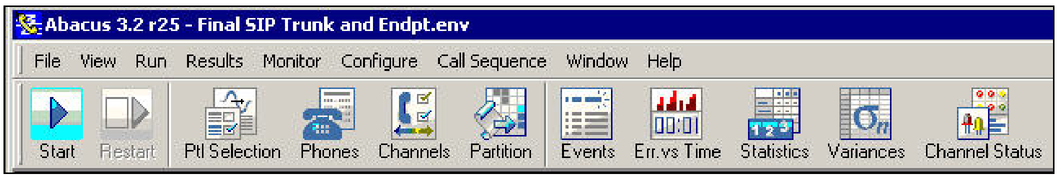
7.2. Verify Avaya SIP Enablement Services

From the Linux shell of SES, use the "trustedhost –L" command to verify the IP address of the Abacus 5000 originating ICG3 card is listed as a trusted host.

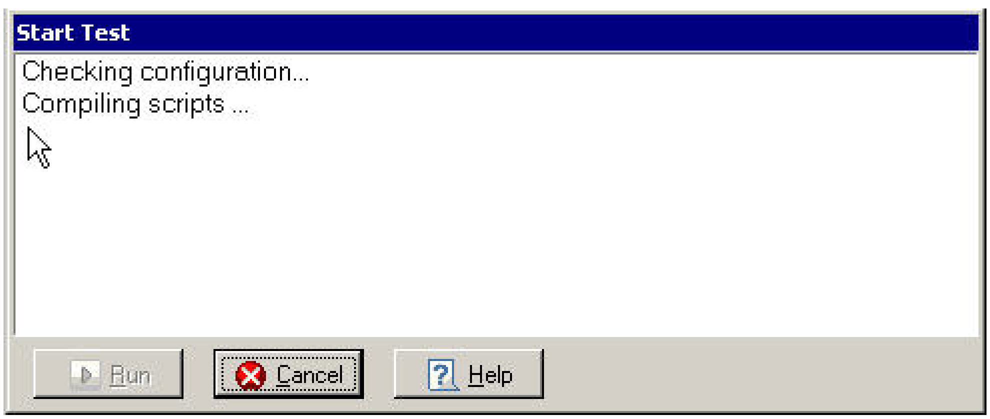
```
craft@CCS-DevCon1> trustedhost -L
Third party trusted hosts.
       Trusted Host          |       CCS Host Name        |         Comment
--------------------------+---------------------------+-------------------------
192.45.61.118             | CCS-DevCon1.devconnect.com |
192.45.30.151             | CCS-DevCon1.devconnect.com |
```

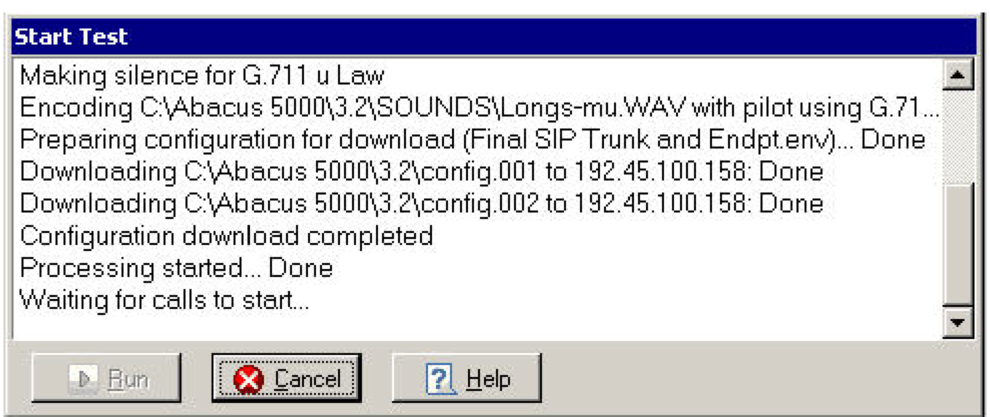
7.3. Verify Abacus 5000

Click on the **Start** icon from the main menu bar.

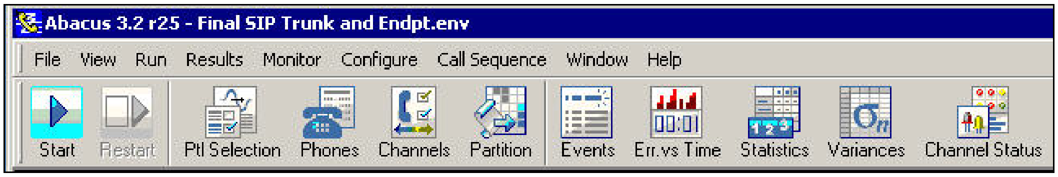


Verify that the scripts can be compiled successfully without any errors, as shown in the screens below.

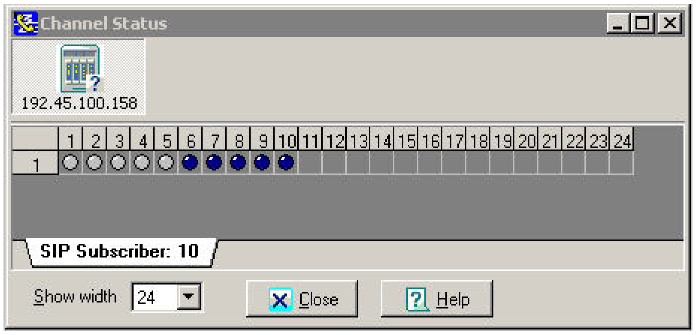




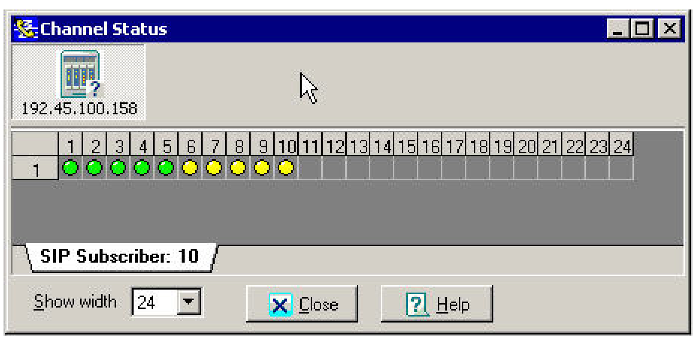
Verify the channel status by clicking on the **Channel Status** icon from the main menu bar.



The **Channel Status** screen will have the terminating channels, in this case channels 6-10, in navy color upon initial registration with Avaya SES.



When the SIP calls have been established, then the color code for the corresponding channel will become green for the originating channels, and yellow for the terminating channels, as shown below. In this case, channels 1-5 are the originating channels, and channels 6-10 are the terminating channels.



Verify the statistics by clicking on the **Statistics** icon from the main menu bar. In the **Statistics** screen displayed below, look for **Script completions**, **Registration Successes**, and the absence of any **Errors**. The **Registration Retry Attempts** field will always be non-zero, as the registration process is to authenticate before registration.

Statistics

192.45.100.158

	Total	Orig	Term	1	2	3	4	5	6	7	8	9	10
Script attempts	625	625	0	125	125	125	125	125	0	0	0	0	0
Script completions	625	625	0	125	125	125	125	125	0	0	0	0	0
% Script completions	100.0	100.0	0.00	100.0	100.0	100.0	100.0	100.0	0.00	0.00	0.00	0.00	0.00
Call attempts	1250	625	625	125	125	125	125	125	125	125	125	125	125
Call completions	1250	625	625	125	125	125	125	125	125	125	125	125	125
% Call completions	100.0	100.0	100.0	100.0	100.0	100.0	100.0	100.0	100.0	100.0	100.0	100.0	100.0
Call attempts per second	0.21	0.11	0.11	0.021	0.021	0.021	0.021	0.021	0.021	0.021	0.021	0.021	0.021
Registration Successes	10	0	10	0	0	0	0	0	2	2	2	2	2
Registration Failures	0	0	0	0	0	0	0	0	0	0	0	0	0
Registration Retry Attempts	10	0	10	0	0	0	0	0	2	2	2	2	2
Registration In Progress	0	0	0	0	0	0	0	0	0	0	0	0	0
Errors	0	0	0	0	0	0	0	0	0	0	0	0	0

SIP Subscriber: 10

Verify the variances by clicking on the **Variances** icon from the main menu bar. In the **Variances** screen displayed below, look for **PESQ** scores. PESQ scores range from -0.5 to 4.5, where 4.5 indicates there is no perceptible difference between the speech sample and the degraded signal. The perfect PESQ scores below were accomplished with direct media shuffling and G.711 audio codec. The scores may be lower for scenarios with non-media shuffling and use of audio codec that requires compression.

Variances

192.45.100.158

	Count	Minimum	Average	Maximum
Acknowledgement (s)	635	0.765	0.793	0.815
Call length terminate (s)	630	36.458	36.499	36.786
Call length originate (s)	630	36.414	36.473	36.770
RTP Jitter (msec)	18112	0	0	2
Tear Down (msec)	630	28	42	340
PESQ - All files	14535	4.500	4.500	4.500
PESQ - Longs-mu.WAV	14535	4.500	4.500	4.500
PESQ-LQ - All files	14535	4.500	4.500	4.500
PESQ-LQ - Longs-mu.WAV	14535	4.500	4.500	4.500
R-Factor - All files	14535	93.2	93.2	93.2
R-Factor - Longs-mu.WAV	14535	93.2	93.2	93.2
JMOS - All files	14535	3.934	3.934	3.934
JMOS - Longs-mu.WAV	14535	3.934	3.934	3.934
RTP Packet Loss (per RTCP packet)	18112	0	0	0
RTP Packets Out Of Order (per RTCP packet)	10728	0	100	269
RTP Packets Late Arrival (per RTCP packet)	10728	0	0	0

8. Support

Technical support on Spirent Abacus 5000 can be obtained through the following:

- Email the Spirent support center via support@spirentcom.com.
- Call the Spirent support center at 1-800-SPIRENT.

9. Conclusion

These Application Notes describe the configuration steps required for Spirent Abacus 5000 Release 3.2 to successfully interoperate with Avaya Communication Manager 3.0.1 and Avaya SIP Enablement Services 3.0. All feature and serviceability test cases were completed successfully.

There were two observations from the compliance testing. The first is any customized setting of the Protocol Selection SUT will not be preserved in the environment file. The workaround is to manually change the "SIP Default" value corresponding to the second ICG3 card back to the custom SUT, upon each loading of the environment file.

The second observation is that during a test run, when the LAN cable is pulled from the ICG3 card for longer than 30 seconds and then restored, no further calls can be completed. The workaround is to manually stop and restart the test run.

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, June 2005, available at http://support.avaya.com

- *SIP Enablement Services R3.0 Installation and Administration Guide*,, Issue 5.1, July 2005, available at http://support.avaya.com

- *Abacus 5000 IP Telephony Migration Test System*, available from the Spirent Abacus 5000 Version 3.2 Installation CD.

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.