Avaya Solution & Interoperability Test Lab

Application Notes for SER TSP500 with Avaya Communication Manager using Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for the SER TSP500 predictive dialing switch to successfully interoperate with Avaya Communication Manager using Avaya SIP Enablement Services. The SER TSP500 is a predictive dialing switch that supports VoIP, and integration with Avaya Communication Manager is achieved through Avaya SIP Enablement Services with dedicated SIP trunk connections to available agents.

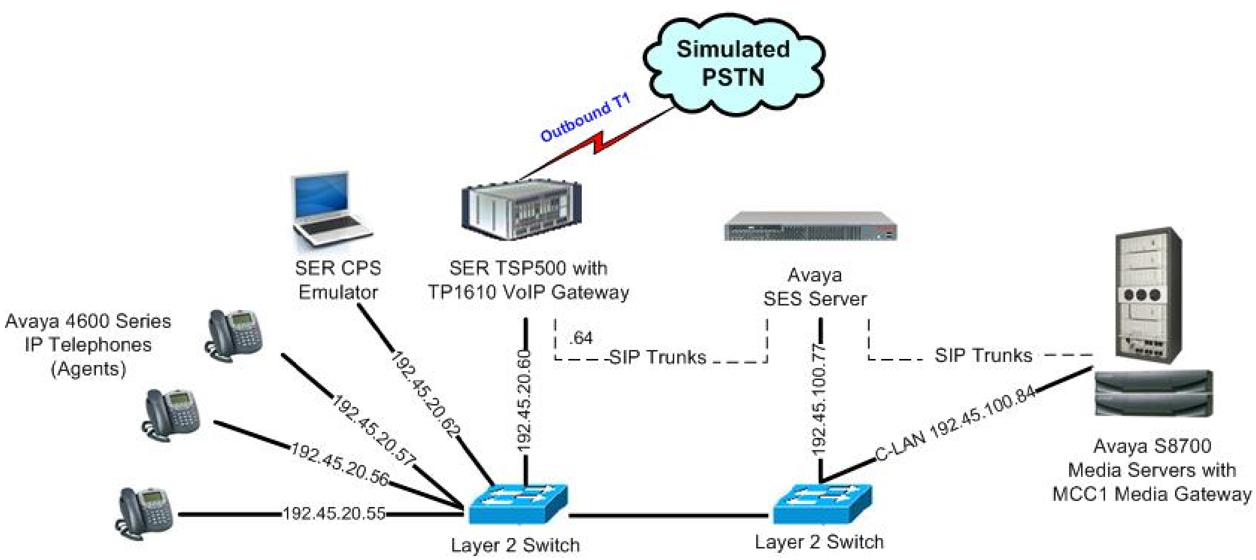
Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

The SER TSP500 is a predictive dialing switch that supports VoIP. The integration with Avaya Communication Manager is achieved through Avaya SIP Enablement Services (SES), with dedicated SIP trunk connections from TSP500 to available agents.

Agents are administered as regular station users on Avaya Communication Manager, and have desktop interfaces to the SER Call Processing System (CPS) application to initiate all Automatic Call Distribution (ACD) and call related actions, such as login/logout, change work modes, and hang up calls. Note that the CPS application provides all ACD functionality in this scenario.

The CPS application typically runs on a separate server from TSP500. For the compliance testing, a laptop with serial console connection to TSP500 is used to emulate the CPS application requests via the command line interface. The Avaya 4600 Series IP Telephones running the H.323 protocol are used as agent telephones. The focus of the compliance testing is on the SIP trunk interface between TSP500 and Avaya SES. The configuration used for the compliance testing is shown below.



When an agent is available to receive outbound calls, the CPS application sends a request to TSP500 to establish a dedicated audio path connection to the agent. This audio path connects a port from the SIP-compliant TP1610 VoIP Gateway Card on TSP500 to the agent on Avaya Communication Manager. The connection request appears as an incoming SIP trunk call to Avaya SES, and is routed via separate SIP trunks to Avaya Communication Manager. The SIP trunk connection will stay up for as long as the agent is available to handle outbound calls.

The TSP500 then utilizes an algorithm to place outbound calls via its NMS CG6500C T1/E1/DSP Resource Card to the PSTN, with call progress tones and tone detections all handled by TSP500. When an outbound call has been classified to be answered, then the TSP500 internal switch fabric connects the answered call to the audio path of the agent.

2. Equipment and Software Validated

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

Equipment	Software
Avaya S8700 Media Servers	Avaya Communication Manager 3.1.2, load 632.1
Avaya MCC1 Media Gateway • TN799DP C-LAN Circuit Pack	 HW01 FW017
Avaya SIP Enablement Services	3.1, load 18
Avaya 4610SW IP Telephones (H.323)	2.1
SER TSP500 Server • TP1610 VoIP Gateway Card • NMS CG6500c T1/E1/DSP Resource Card	 4.6 NA

3. Configure Avaya Communication Manager

The procedures for configuring 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 agents

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 trunks. Navigate to **Page 2**, and compare the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections to be launched by SER TSP500.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 100   50
           Maximum Concurrently Registered IP Stations: 100   4
             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   15

   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 TSP500. Select an audio codec type in the **Audio Codec** field, in this case “G.729”. The actual codec set number and codec type may vary. **Section 5.3** contains a table listing of the audio codec types that successfully interoperated between TSP500 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.729            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 TSP500. Enter the audio codec set number from the **IP Codec Set** screen above into the **Codec Set** field. For the **Authoritative Domain** field, enter the SIP domain name of the SES server from **Section 4.1**. Enable the **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct** fields to allow for audio shuffling. 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: devconnect.com
    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: 3048
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 “192.45.100.77” 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 .100.77                            .   .   .

( 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 by using 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
```

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. Submit these changes.

```
change 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
```

3.8. Administer Agents

Administer an agent telephone by using 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 other station types may be used for the agents. The compliance testing utilized the "4610" station type, and the agents are registered to the same C-LAN configured in **Section 3.4**.

- **Type:** "4610"
- **Port:** "IP"
- **Name:** Enter a descriptive name.
- **Security Code:** Enter a desired security code.

```
add station 22991                                              Page   1 of   4
                                     STATION

Extension: 22991                           Lock Messages? n          BCC: 0
     Type: 4610                            Security Code: 123456     TN: 1
     Port: IP                            Coverage Path 1:            COR: 1
     Name: TSP 500 Agent x22991          Coverage Path 2:            COS: 1
                                         Hunt-to Station:

STATION OPTIONS
              Loss Group: 19             Personalized Ringing Pattern: 1
                                                     Message Lamp Ext: 22991
            Speakerphone: 2-way                   Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal                  Media Complex Ext:
   Survivable Trunk Dest? y                              IP SoftPhone? n

                                                  Customizable Labels? Y
```

Repeat the "add station n" command to add the desired number of agent telephones. For the compliance testing, three agent telephones were administered as shown below.

```
list station 22991 count 3

                                  STATIONS

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

22991    S00016   TSP500 Agent x22991                            1
 4610                                 no                         1
22992    S00017   TSP500 Agent x22992                            1
 4610                                 no                         1
22993    S00018   TSP500 Agent x22993                            1
 4610                                 no                         1
```

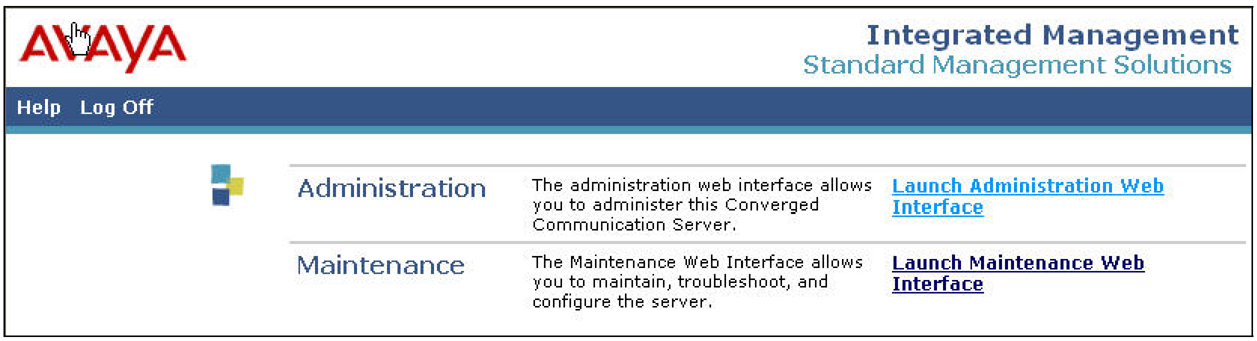
4. Configure Avaya SIP Enablement Services

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

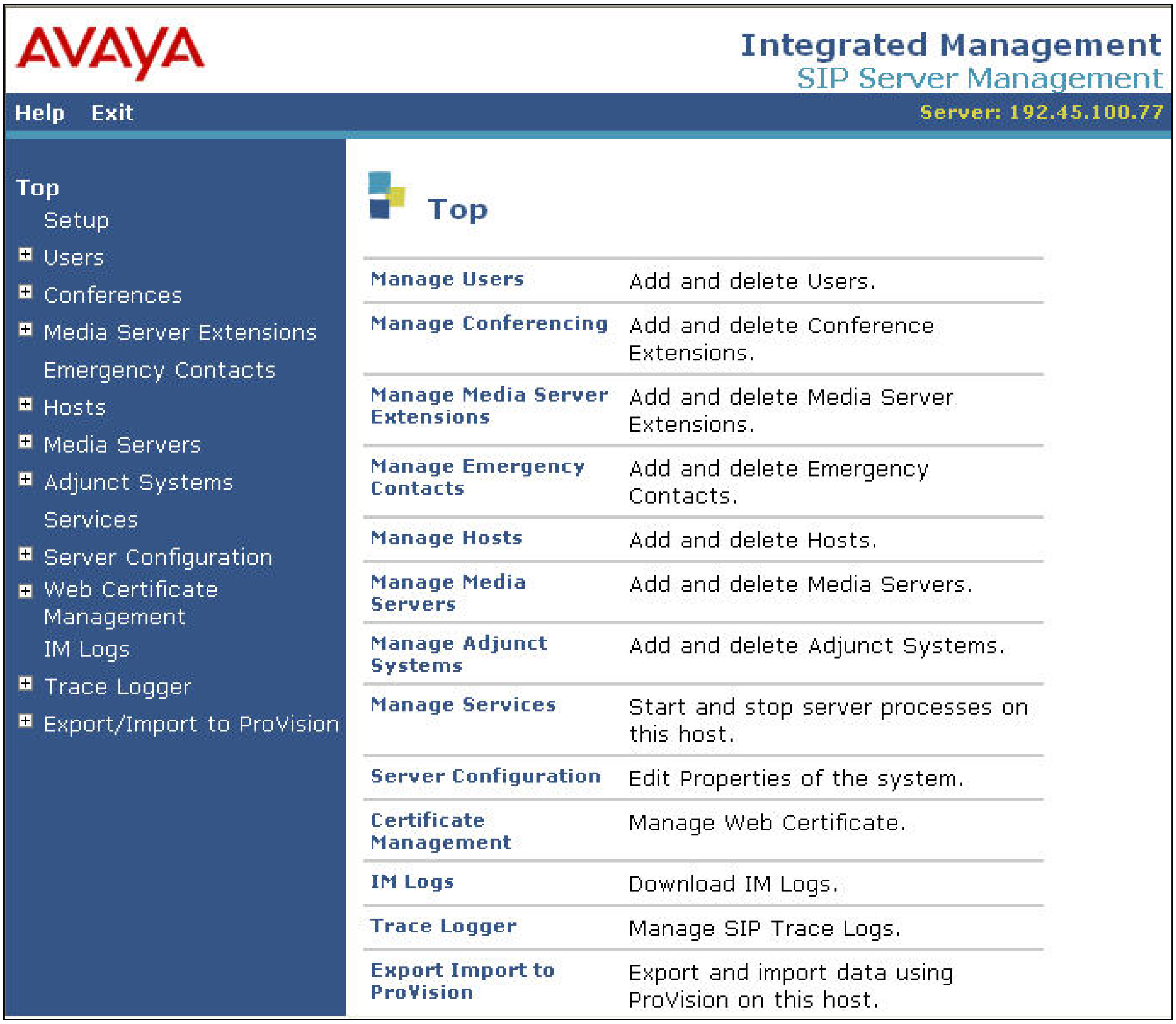
- Obtain home server host
- Administer media server
- Administer media server address map
- Administer trusted host

4.1. Obtain Home Server Host

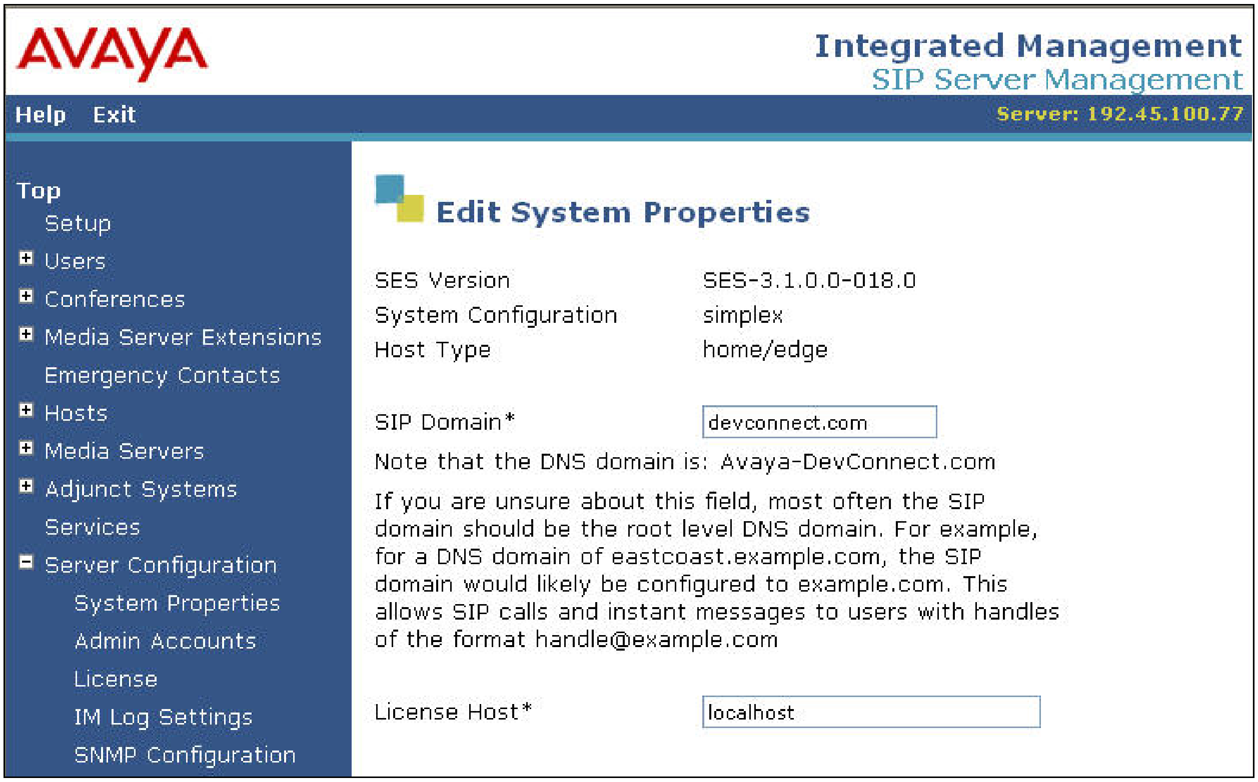
Access the SES administration web interface by using the URL "http://<ip-address>/admin" in an Internet browser window, where "<ip-address>" is the IP address of the SES server. Note that the IP address for the SES server may vary, and in this case "192.45.100.77" is used, as administered in **Section 3.3**. Log in with the appropriate credentials and select the **Launch Administration Web Interface** option.



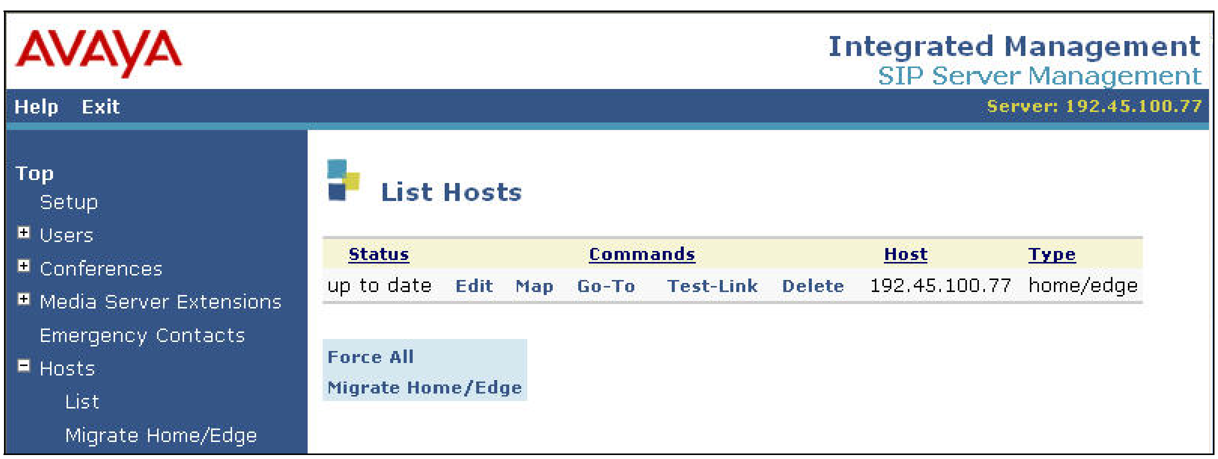
The **Top** screen is displayed next. If this is the initial setup of the SES server, then follow the Avaya SES documentation in **Section 10** to administer the SIP domain and host. These Application Notes assume the SES server has already been configured with the proper domain and host information.



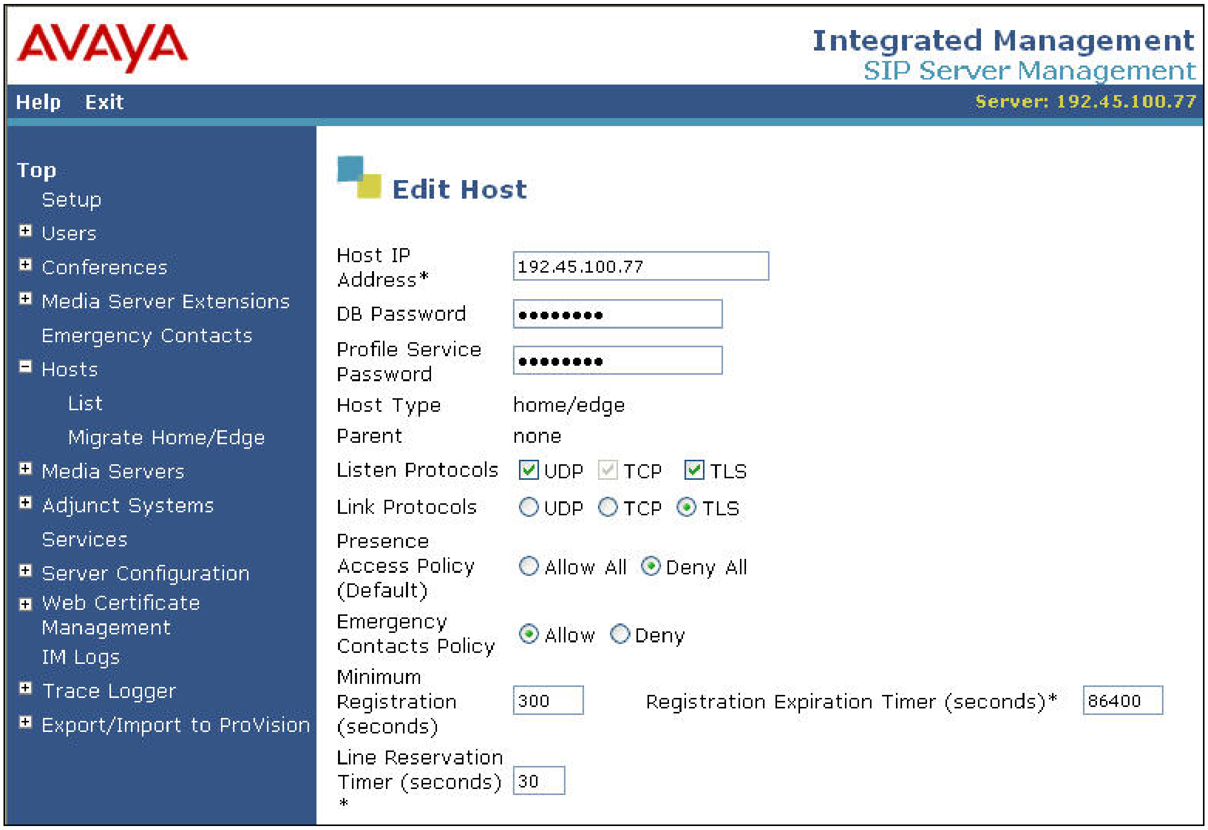
Select **Server Configuration** > **System Properties** from the left pane to display the **Edit System Properties** screen below. Make a note of the value in the **SIP Domain** field, in this case "devconnect.com".



Select **Hosts** > **List** from the left pane to display the **List Hosts** screen. Click on the **Edit** button for each host to find the host serving as the home server.



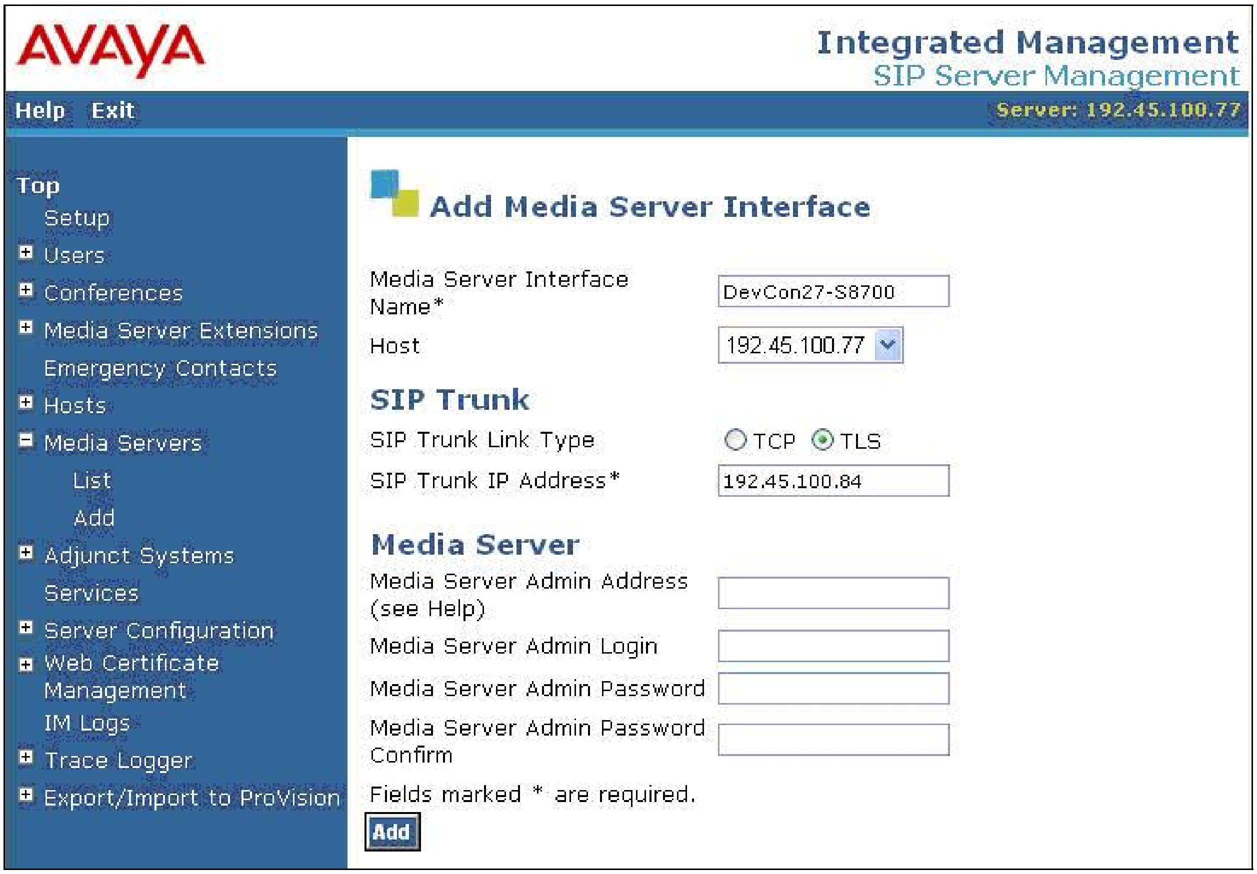
In the **Edit Host** screen, check the value of the **Host Type** field. For the compliance testing, only one host is administered as both the edge and home server, as indicated by the "home/edge" value in the **Host Type** field shown below. The IP address of this home server is "192.45.100.77", indicated in the **Host IP Address** field. This will be used to configure the media server interface.



4.2. Administer Media Server

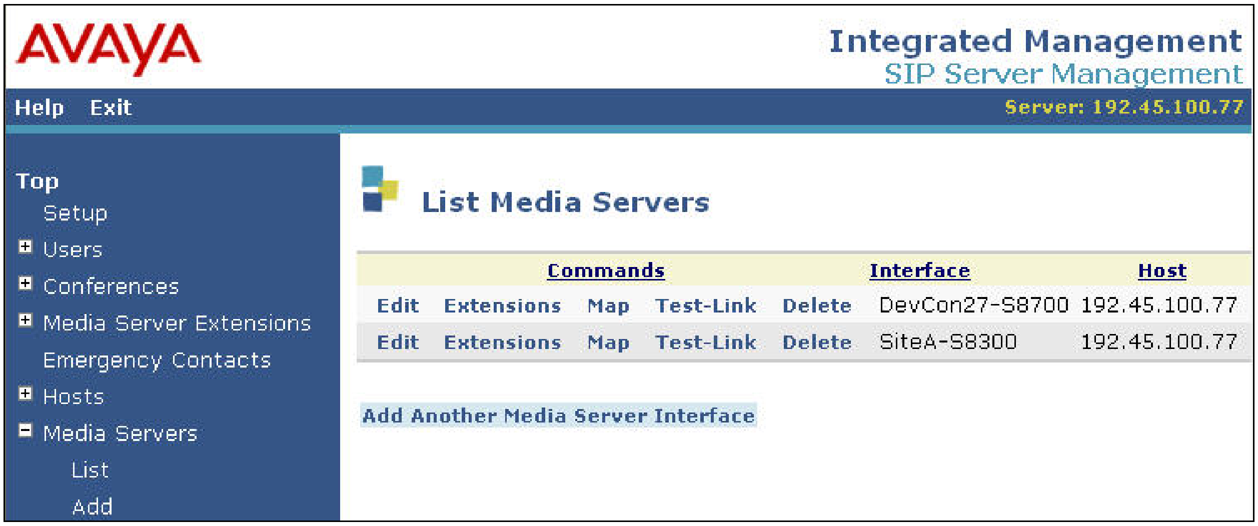
Select **Media Servers > Add** from the left pane to display the **Add Media Server Interface** screen. This screen associates a media server with a host. Enter the following values for the specified fields, and retain the default values for the remaining fields. Click on **Add** in the lower right pane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

- **Media Server Interface Name:** A descriptive name.
- **Host:** Select the IP address 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 Media Server Address Map

Select **Media Servers > List** from the left pane to display the **List Media Servers** screen below. Click on the **Map** link associated with the media server interface administered in **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 to add a media server address map.

The **Add Media Server Address Map** screen is displayed next. This screen is used to specify which calls are 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, a **Pattern** of "^sip:2[0-9]{4}" was used to match to any extensions in the range of 20000-29999. Maintain the check in the **Replace URI** field, 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 the **Add** button.

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.

4.4. Administer Trusted Host

Administer TSP500 as a trusted host, so that the SIP messages from TSP500 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 TP1610 VoIP Gateway Card from **Section 5.1**, and "y" is the host name or IP address of the SES home server from **Section 4.1**.

```
craft@SES-DevCon1> trustedhost -a 192.45.20.64 -n 192.45.100.77
192.45.20.64 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 SER TSP500

These application notes assume the SER TSP500 has already been configured with the basic outbound trunks, "remote" agents with VoIP, and default routing information. The agents need to be configured as "remote" as they reside physically on Avaya Communication Manager and not local to TSP500. This section provides the procedures for configuring the TP1610 VoIP Gateway Card on TSP500. The procedures include the following areas:

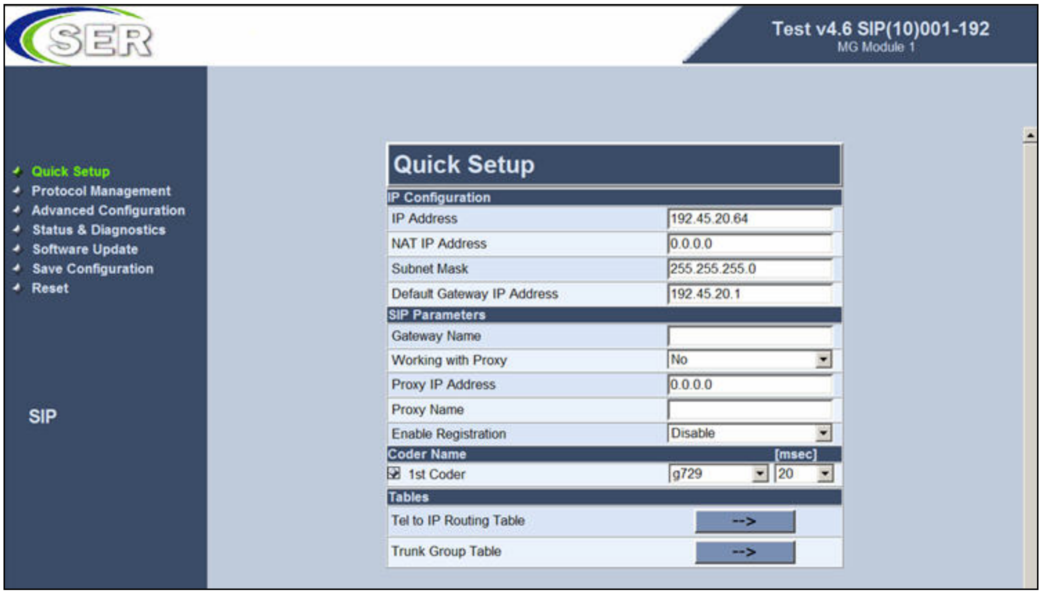
- Verify IP addresses
- Administer general parameters
- Administer coders
- Administer DTMF & dialing
- Administer telephone to IP routing
- Administer trunk group table

The configuration of the CPS application and agent interfaces are performed by SER technicians and partners, and are outside the scope of these Application Notes.

5.1. Verify IP Addresses

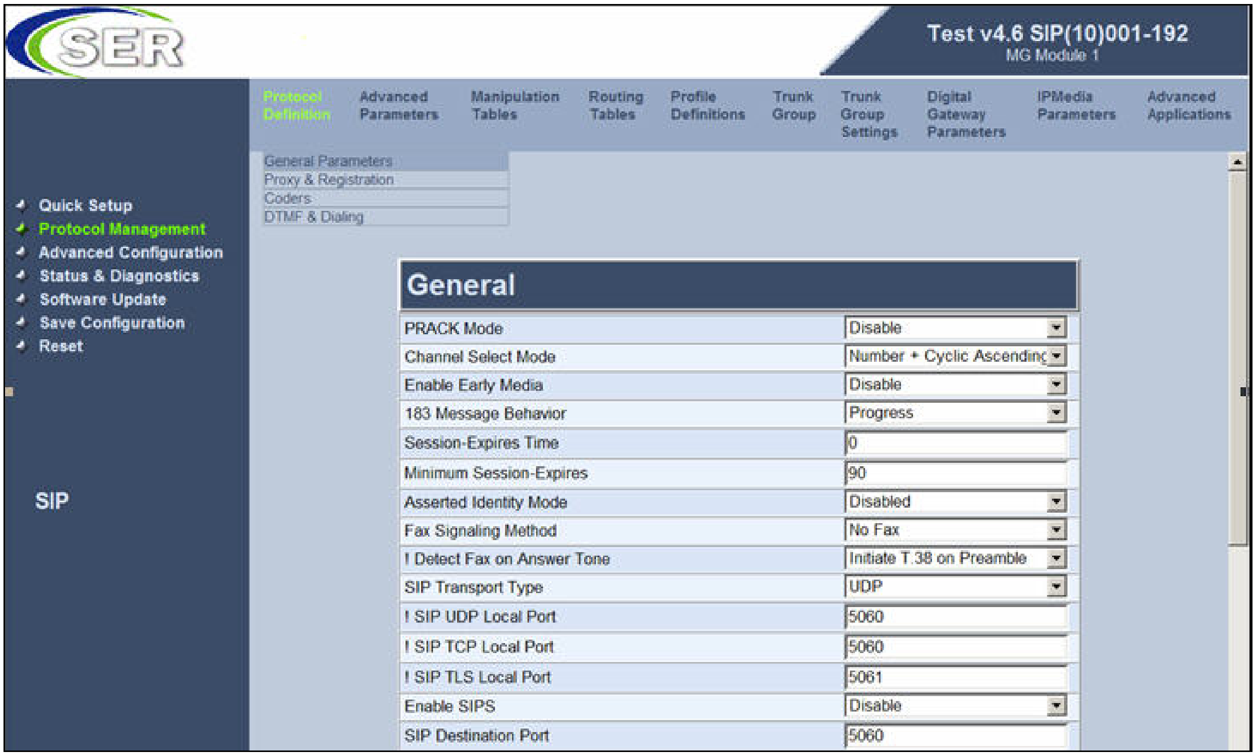
Access the TP1610 VoIP Gateway Card administration web interface by using the URL "http://<ip-address>" in an Internet browser window, where "<ip-address>" is the IP address of the TP1610 VoIP Gateway Card. Note that the IP address for the gateway card may vary, and in this case "192.45.20.64" is used, and was configured as part of installation. Log in with the appropriate credentials and select the **Quick Setup** option.

The **Quick Setup** screen is displayed, as shown below. Verify the values in the **IP Address**, **Subnet Mask**, and **Default Gateway IP Address** fields. These values were entered as part of the TP 1610 VoIP Gateway Card installation.



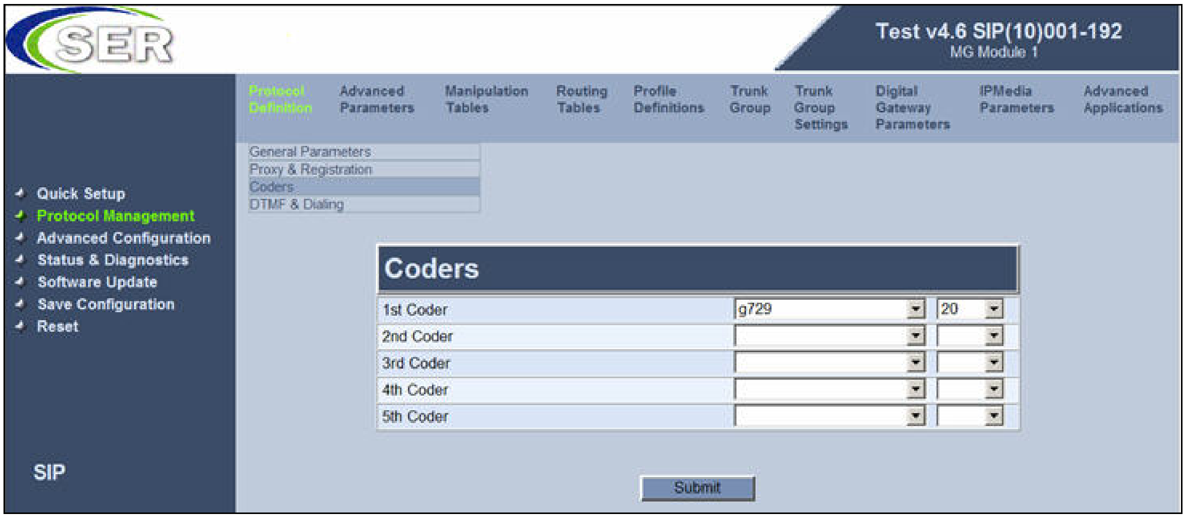
5.2. Administer General Parameters

Select **Protocol Management** from the left pane, followed by **Protocol Definition > General Parameters** in the right pane. The **General** screen is displayed, as shown below. Select "UDP" from the **SIP Transport Type** drop down list. Enter "5060" for the **SIP UDP Local Port** and **SIP Destination Port** fields, and retain the default values for all remaining fields. Scroll down to the bottom of the screen to click the **Submit** button (not shown below).



5.3. Administer Coders

Select **Protocol Definition > Coders** in the right pane to display the **Coders** screen. Select the desired audio codec type and packet size from the drop down lists, and click **Submit**. Note that the selected audio codec type needs to match the audio codec type administered on Avaya Communication Manager in **Section 3.2**.

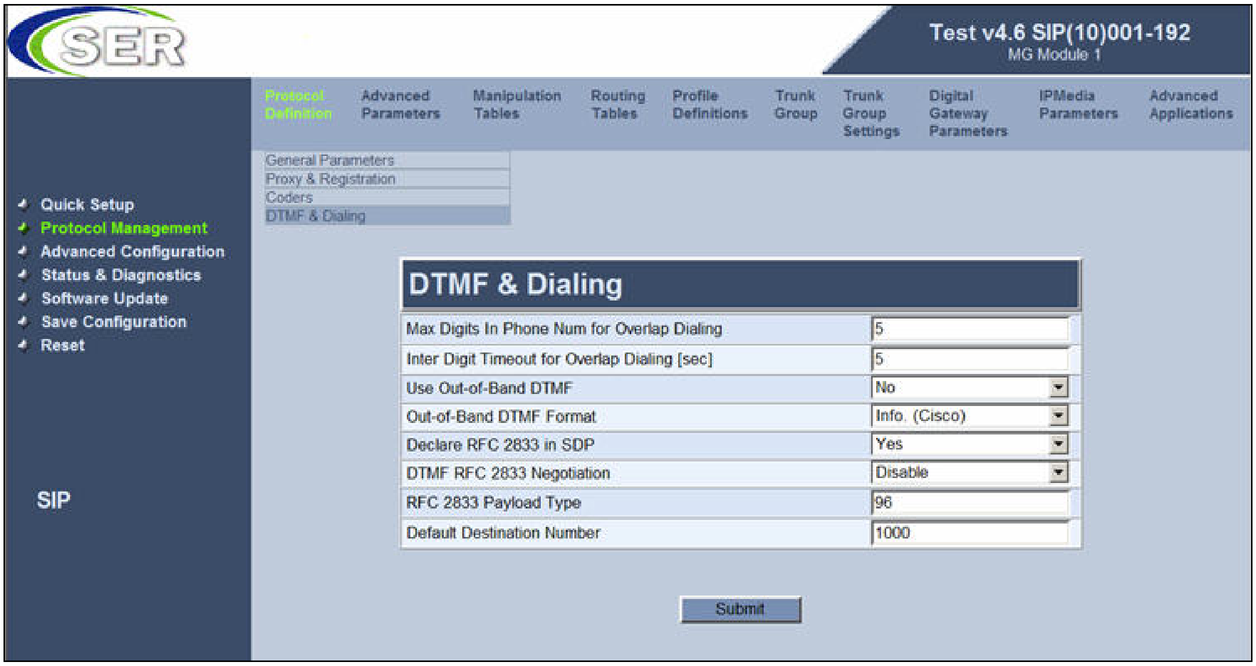


The following are the audio codec types that successfully interoperated between SER TSP500 and Avaya Communication Manager during the compliance testing.

SER TSP500	Avaya Communication Manager
g711Ulaw64k	G.711MU
g729	G.729
g729	G.729A
g729_AnnexB	G.729B

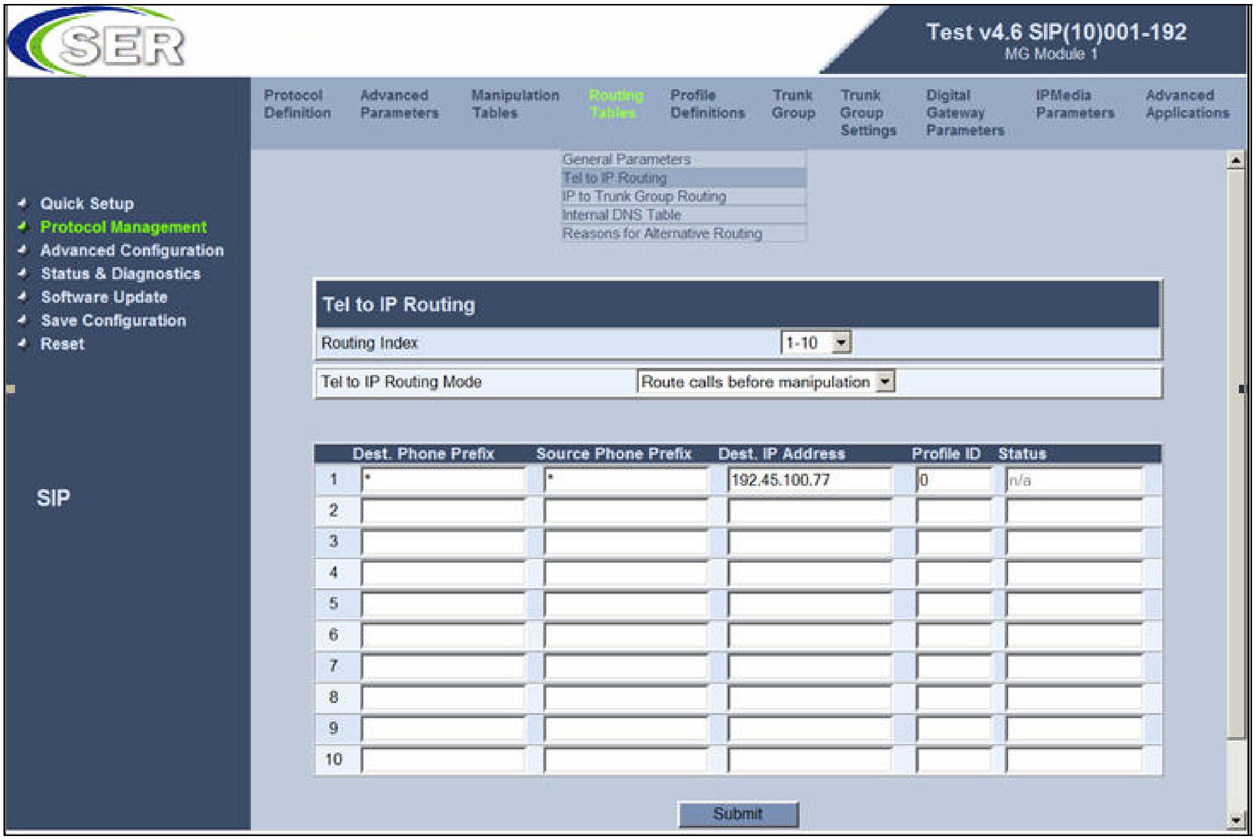
5.4. Administer DTMF & Dialing

Select **Protocol Definition > DTMF & Dialing** in the right pane to display the **DTMF & Dialing** screen. For the **Max Digits in Phone Num for Overlap Dialing** field, enter the maximum number of digits used for agent extensions. In this case, the maximum number of digits is "5", as shown below. Retain the default values in all other fields, and click **Submit**.



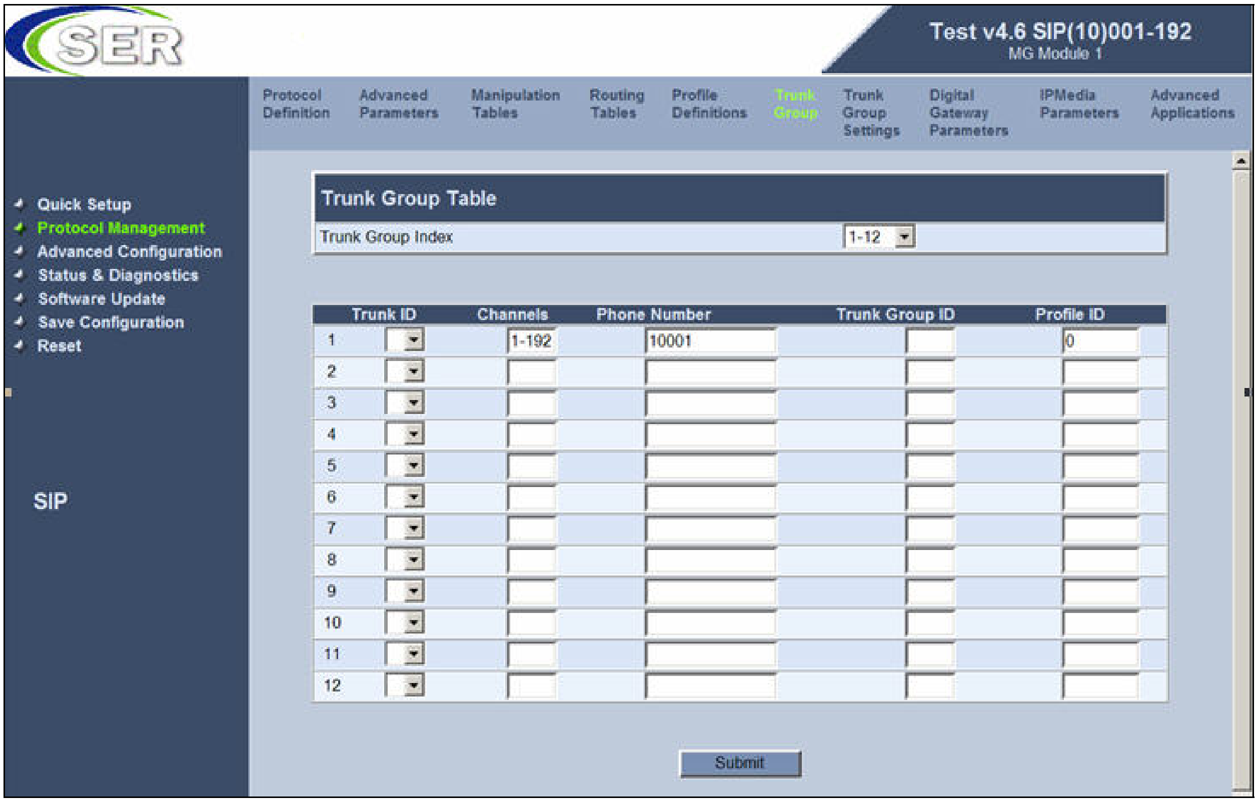
5.5. Administer Telephone to IP Routing

Select **Routing Tables > Tel to IP Routing** in the right pane to display the **Tel to IP Routing** screen. Maintain the default values in the **Routing Index** and **Tel to IP Routing Mode** fields. Enter a table entry as shown below, to enable routing of any dialed numbers to Avaya SES. For the **Dest. IP Address** field, enter the IP address of the Avaya SES server from **Section 3.3**. Retain the default values in the other fields, and click **Submit**.



5.6. Administer Trunk Group Table

Select **Trunk Group** in the right pane to display the **Trunk Group Table** screen. This screen is used to define the trunks and logical telephone numbers. Maintain the default value in the **Trunk Group Index** field, and enter a table entry as shown below. The entry created for the compliance testing defined a trunk group with “192” channels, using sequential telephone numbers starting with “10001”. The sequential telephone numbers are logical numbers assigned to the range of channels, and note that the actual numbers may vary. Click **Submit**.



6. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing focused on verifying SER TSP500 establishment of dedicated SIP trunk connection to agents, and connection of answered outbound calls to the agents. Various audio codec were used with and without media shuffling. Testing also included rainy day scenarios to verify TSP500 handling of error conditions, such as invalid agent extension and agent ring no answer.

The serviceability testing focused on verifying the ability of TSP500 to recover from adverse conditions, including disconnecting the Ethernet cable to TSP500 and Avaya SES, and busying out the SIP signaling group on Avaya Communication Manager.

6.1. General Test Approach

All tests were performed manually. The laptop with a serial console connection to TSP500 was used to emulate the requests from the CPS application for agent login/logout, agent drop, outbound call placement, and connection of answered outbound call to available agent.

Traces of SIP messages were captured on both the SER TP1610 VoIP Gateway Card and Avaya SES server for trouble shooting and verification of scenarios.

6.2. Test Results

All test cases were executed and passed.

7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Communication Manager, Avaya SES, and SER TSP500.

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 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 SER TP1610 VoIP Gateway Card and agent telephone. 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. 45.100. 77 : 5061

G.729       Audio:         192. 45. 20. 64  : 6240    192. 45. 20. 55 : 2744
            Video:
      Video Codec:
                                            Authentication Type: None
    Audio Connection Type: ip-direct
```

Verify the status of a connected agent by using the “status station n”, where “n” is the extension of an administered agent telephone from **Section 3.8**. Verify the **Service State** is “in-service/ off-hook”, as shown below.

```
status station 22991                                           Page   1 of   6
                              GENERAL STATUS
     Administered Type: 4610              Service State: in-service/off-hook
        Connected Type: 4610           TCP Signal Status: connected
             Extension: 22991
                  Port: S00016        Parameter Download: complete
           Call Parked? no                 SAC Activated? no
      Ring Cut Off Act? no            CF Destination Ext:
Active Coverage Option: 1
```

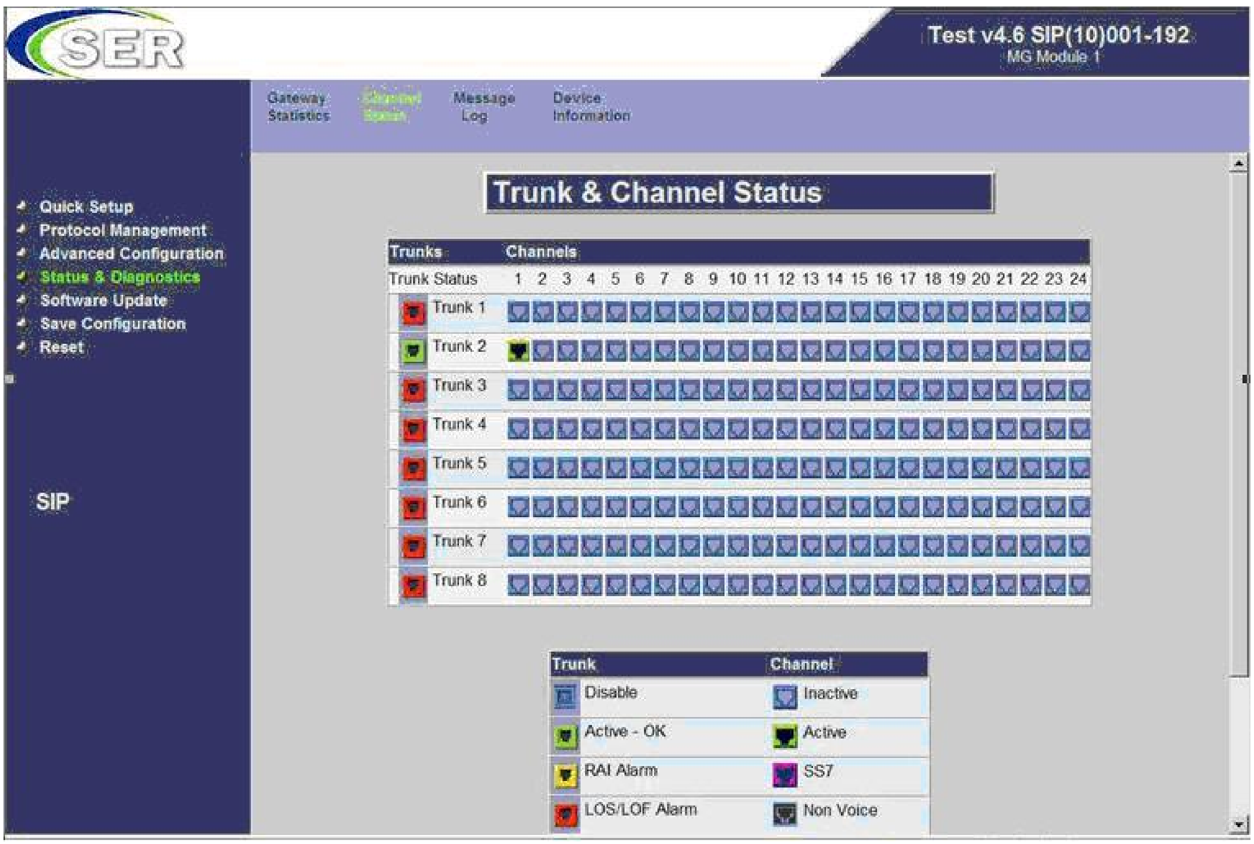
7.2. Verify Avaya SIP Enablement Services

From the Linux shell of SES, use the “trustedhost –L” command to verify the IP address of the SER TP1610 VoIP Gateway Card is listed as a trusted host.

```
craft@SES-DevCon1> trustedhost -L
Third party trusted hosts.
       Trusted Host        |       CCS Host Name        |         Comment
--------------------------+---------------------------+-------------------------
192.45.20.64              | 192.45.100.77              |
```

7.3. Verify SER TSP500

Verify the status of a connected SIP trunk by selecting **Status & Diagnostics** from the left pane, followed by **Channel Status** in the right pane. Verify that there is an active trunk, as shown below.



8. Support

Technical support on SER TSP500 can be obtained through the following:

- **Phone:** (800) 274-5676
- **Email:** info@ser.com

9. Conclusion

These Application Notes describe the configuration steps required for SER TSP500 3.3 to successfully interoperate with Avaya Communication Manager 3.1.2 using Avaya SIP Enablement Services 3.1.

10. Additional References

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

- *Administrator Guide for Avaya Communication Manager*, Document 03-300509, Issue 2, February 2006, available at http://support.avaya.com.

- *Installing and Administering SIP Enablement Services R3.1*, Document ID 03-600768, Issue 1.4, February 2006, available at http://support.avaya.com.

- *SIP Support in Release 3.1 of Avaya Communication Manager Running on the S8300, S8400, S8500 series, and S8700 series Media Server*, Document 555-245-206, Issue 6, February 2006, available at http://support.avaya.com.

- *Mediant 2000, Mediant 1000, TP-1610 and TP-260 SIP User's Manual*, Version 4.6, Document #LTRT-68803, available on TP1610 VoIP Gateway Card 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.