Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between Taiwan Fixed Network SIP Trunking Service and an Avaya IP Telephony Solution – 1.0

Abstract

These Application Notes describe the steps to configure SIP trunking between Taiwan Fixed Network (TFN) SIP Trunking service and an Avaya IP Telephony solution. The Avaya solution consists of Avaya SIP Enablement Services, Avaya Communications Manager, and various Avaya SIP, H.323, digital and analog end points.

Taiwan Fixed Network (TFN) is the second largest operator in Taiwan and is partnering with Avaya to promote their new "all communications in one pipe" SIP trunk service. It is based on Metro Ethernet technology and uses Huawei softswitch. It allows TFN to utilize the features of Avaya Communication Manager to provide enterprise wide services.

Taiwan Fixed Network is a Service Provider member of the Avaya Developer*Connection* program. Information in these Application Notes has been obtained through Developer*Connection* compliance testing and additional technical discussions. Testing was conducted remotely via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab in Singapore.

1. Introduction

These Application Notes describe the steps for configuring SIP trunking between the TFN SIP Trunking Service and an Avaya IP telephony solution consisting of Avaya SIP Enablement Services, Avaya Communication Manager and various Avaya telephony endpoints. These endpoints included IP telephones (using SIP and H.323 protocols), traditional analog and digital phones.

Taiwan Fixed Network (TFN) is the second largest operator in Taiwan and is partnering with Avaya to promote their new "all communications in one pipe" SIP trunk service. It is based on Metro Ethernet technology and uses a Huawei softswitch. It allows TFN to utilize the features of Avaya Communication Manager to provide enterprise wide services.

Figure 1 illustrates a sample Avaya IP telephony solution connected to TFN's SIP trunking service. This is the configuration used during the Developer*Connection* compliance testing process.

The Avaya IP telephony solution used to create a simulated customer site contained:

• Avaya S8300B Media Server with an Avaya G700 Media Gateway. The S83000B served as the host processor for Avaya Communication Manager.

• Avaya SIP Enablement Services (SES) software operating on an Avaya S8500C server platform.

• Avaya 4600 series IP telephones (configured to use either the SIP or H.323 protocol), Avaya 9600 series H.323 IP Telephones, Avaya Digital (DCP) telephones, and Avaya 6200 series analog telephones.

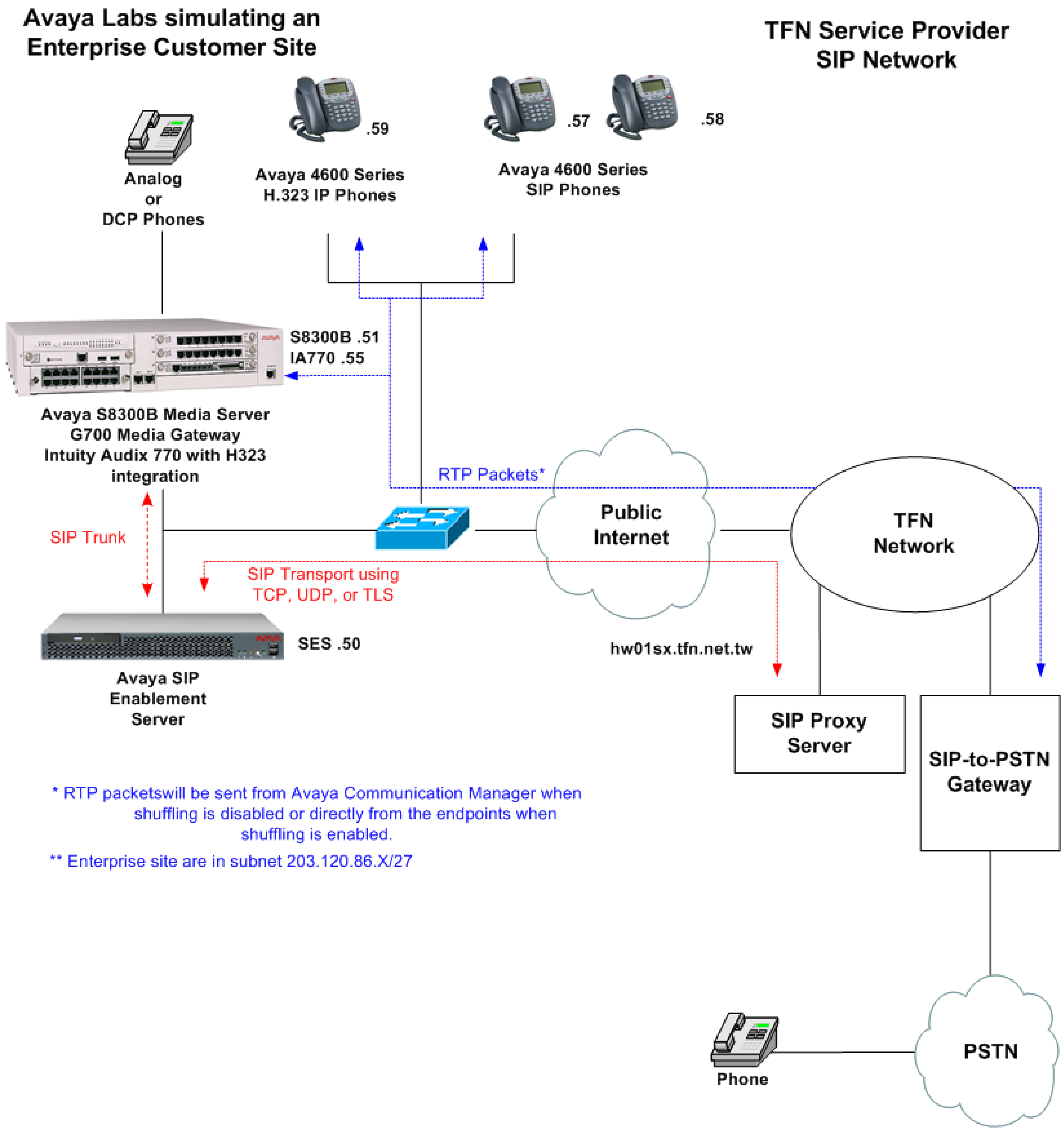

Figure 1: Avaya IP Telephony Network using TFN SIP Trunking Service

1.1 Call Flows

To better understand how calls are routed between the PSTN and the enterprise site shown in **Figure 1** using SIP trunks, two call flows are described in this section. The first call scenario illustrated in **Figure 2** is a PSTN call to the enterprise site terminating on a typical analog telephone supported by Avaya Communication Manager.

1. A user on the PSTN dials a TFN provided DID number assigned to an Avaya Communication Manager telephone at the enterprise site. The PSTN routes the call to the TFN network (as the local service provider) which routes the DID number to the assigned customer.

2. Based on the DID number, TFN offers the call to Avaya SES using SIP signaling messages sent over the converged access facility. Note that the assignment of the DID number and the address of the Avaya SES server was previously established during the ordering and provisioning of the service.

3. Avaya SES routes the call to the Avaya S8300B Media Server running Avaya Communication Manager over a SIP trunk.

4. Avaya Communication Manager terminates the call to the directly connected analog phone as shown in **Figure 2** (step 4). The same process occurs for calls to Avaya digital and H.323 IP phones.

- or –

4a. Inbound calls destined for a SIP extension at the enterprise are routed to Avaya Communication Manager which then transmits the appropriate SIP signaling via Avaya SES to the SIP telephone (as shown by the 4a arrow.)

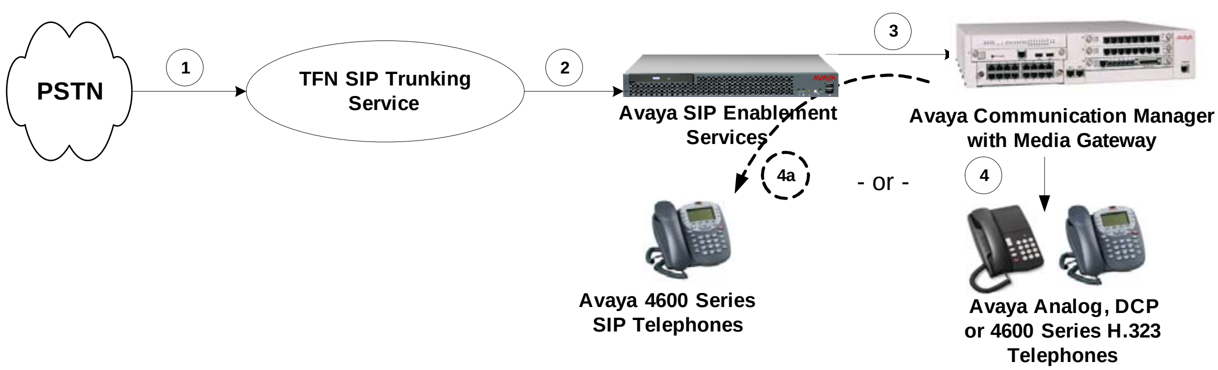

Figure 2: Incoming PSTN Calls to Avaya Communication Manager

Appendix A illustrates an example of a SIP INVITE message sent by TFN for an incoming DID call.

The second call scenario illustrated in **Figure 3** is an outgoing call from an Avaya telephone at the enterprise site to the PSTN via the SIP trunk to TFN.

1. An Avaya H.323, analog or digital telephone served by Avaya Communication Manager originates a call to a user on the PSTN.

- or-

1a. An Avaya SIP telephone originates a call that is routed via Avaya SES (as shown by the 1a arrow) to Avaya Communication Manager.

2. The call request is handled by Avaya Communication Manager where origination treatment such as class of service restrictions and automatic route selection is performed. Avaya Communication Manager selects the SIP trunk and sends the SIP signaling messages to Avaya SIP Enablement Services.

3. Avaya SIP Enablement Services routes the call to TFN.

4. TFN completes the call to the PSTN.

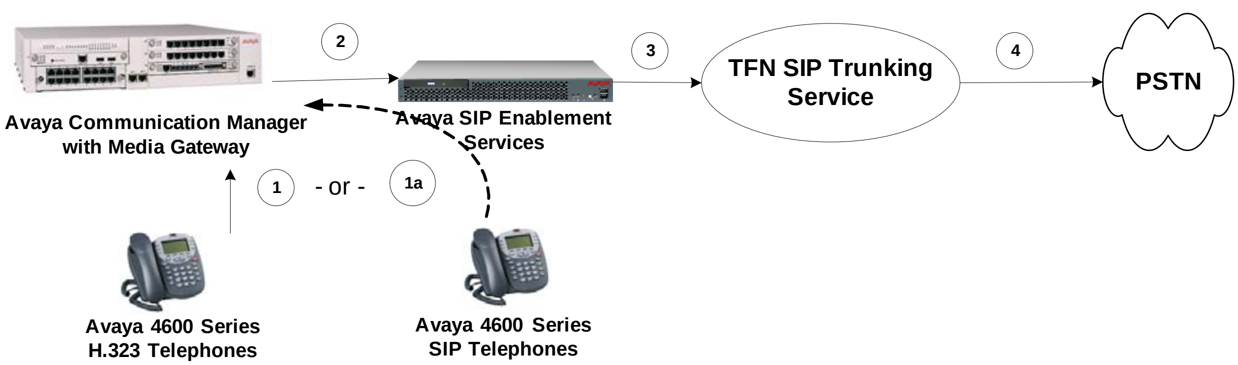

Figure 3: Outgoing Calls from Avaya Communication Manager to the PSTN

2. Equipment and Software Validated

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

Avaya SIP Telephony Solution Components	
Component	**Software Version**
Avaya S8300B Media Server	Avaya Communication Manager 3.1.2 (R013x.01.2.632.1-12866)
Avaya G700 Media Gateway	MGP: 25.33.0 VOIP: 65 MM711 Analog: HW31/FW86 MM717 DCP: HW3/FW4
Avaya SIP Enablement Services on S8500B Media Server	SES-3.1.1.0-114.0
Avaya 4620SW SIP Telephones	Release 2.2.2
Avaya 9630 H.323 IP Telephones	Release 1.2
Avaya 6416 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
TFN VoIP Service Components	
Component	**Version**
TFN Huawei Softswitch	SoftX 3000 R006B03D

Table 1: Equipment and Software Tested

The specific configuration above was used for the TFN compatibility testing. Note that this solution will be compatible with all other Avaya Media Server and Media Gateway platforms running similar versions of Avaya Communication Manager and Avaya SIP Enablement Services.

3. Configure the Avaya Communication Manager

This section describes the steps for configuring a SIP trunk on Avaya Communication Manager. The SIP trunk is established between Avaya Communication Manager and Avaya SIP Enablement Services (SES) server. This trunk will carry the SIP signaling sent to the TFN SIP Trunking Service.

This SIP trunk also provides the trunking for SIP endpoint devices such as Avaya 4600 SIP telephones and Avaya one-X Desktop Edition using Avaya Communication Manager in the recommended OPS configuration. Avaya SIP telephones are configured as off-PBX (OPS) stations on Avaya Communication Manager. OPS SIP stations register with Avaya SES but have calling privileges and features provided by Avaya Communication Manager. Avaya

Communication Manager acts as a back-to-back SIP user agent when a SIP phone places or receives a call over a SIP trunk to a service provider.

Note the use of SIP endpoints is optional. The steps discussed in Sections 3.2 and 4.2 describing SIP endpoints administration may be omitted if SIP endpoints are not used. In the Avaya SIP architecture, the Avaya SES acts as a SIP proxy through which all incoming and outgoing SIP messages flow to TFN. There is no direct SIP signaling path between TFN and Avaya Communication Manager or Avaya SIP endpoints.

For incoming calls, the Avaya SES uses media server routing maps to direct the incoming SIP messages to the appropriate Avaya Communication Manager. Once the message arrives at Avaya Communication Manager further incoming call treatment, such as incoming digit translations, class of service restrictions, etc. may be performed.

All outgoing calls to the PSTN are processed within Avaya Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Avaya Communication Manager selects a SIP trunk, the SIP signaling is routed to the Avaya SES. Within the Avaya SES, host address maps direct the outbound SIP messages to the TFN Softswitch.

The dial plan for the configuration described in these Application Notes consists of 10-digit dialing for local and long-distance calls over the PSTN. However, Directory Assistance calls and International calls were not tested. Avaya Communication Manager routes all calls using Automatic Route Selection (ARS), except for intra-switch calls.

Avaya Communication Manager configuration was performed using the System Access Terminal (SAT). The general installation of the Avaya S8300B Media Server with G700 Media Gateway is presumed to have been previously completed and is not discussed here.

3.1 Sip Trunk Configuration

Step 1: Confirm Necessary Optional Features

Login to the Avaya Communication Manager's SAT interface and confirm that sufficient SIP trunk and Off PBX Telephone capacities are enabled. Use the **display system-parameters customer-options** command to determine these values as shown in **Figure 4**. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

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

     G3 Version: V13
       Location: 2                            RFA System ID (SID): 1
       Platform: 7                            RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 900   143
                                      Maximum Stations: 450   107
                              Maximum XMOBILE Stations: 100   0
                    Maximum Off-PBX Telephones - EC500: 100   1
                    Maximum Off-PBX Telephones -   OPS: 100   3
                    Maximum Off-PBX Telephones - SCCAN: 100   0






        (NOTE: You must logoff & login to effect the permission changes.)
```

Figure 4: System-Parameters Customer-Options Form – Page 1

On Page 2, verify that the number of SIP trunks supported by the system is sufficient for the combination of trunks to the TFN network, SIP endpoints and any other SIP trunks used. Note that each SIP OPS telephone on a call with TFN uses two SIP trunks for the duration of the call.

```
display system-parameters customer-options                      Page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 450   6
           Maximum Concurrently Registered IP Stations: 450   1
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 10    0
  Max Concur Registered Unauthenticated H.323 Stations: 10    0
                  Maximum Video Capable H.323 Stations: 10    0
                   Maximum Video Capable IP Softphones: 10    0
                       Maximum Administered SIP Trunks: 100   30

   Maximum Number of DS1 Boards with Echo Cancellation: 10    0
                             Maximum TN2501 VAL Boards: 0     0
                    Maximum G250/G350/G700 VAL Sources: 50    1
           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.)
```

Figure 5: System-Parameters Customer-Options Form – Page 2

Step 2: Assign Node Names

In the **IP Node Names** form, assign the node name and IP address for Avaya SES at the enterprise site. In this case "SES" and "203.120.86.50" are being used, respectively. The SES node name will be used throughout the other configuration screens of Avaya Communication Manager.

Note, this example shows the Avaya S8300B processor address (procr) is used as the SIP signaling interface. If the Avaya IP Telephony solution utilizes an Avaya G650 Media Gateway, a CLAN is used as the SIP signaling interface.

```
change node-names ip                                            Page   1 of   1
                                  IP NODE NAMES
    Name              IP Address          Name              IP Address
default             0   .0   .0   .0                          .   .   .
msgsvr              203.120.86 .55                            .   .   .
procr               203.120.86 .51                            .   .   .
ses                 203.120.86 .50                            .   .   .
                       .   .   .                              .   .   .
                       .   .   .                              .   .   .
                       .   .   .                              .   .   .
                       .   .   .                              .   .   .
                       .   .   .                              .   .   .
                       .   .   .                              .   .   .
                       .   .   .                              .   .   .
                       .   .   .                              .   .   .
                       .   .   .                              .   .   .
                       .   .   .                              .   .   .
                       .   .   .                              .   .   .
                       .   .   .                              .   .   .
( 4  of 4    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
```

Figure 6: IP Nodes Names Form

Step 3: Define IP Network Region

The **IP Network Region** form specifies the parameters used by the SIP trunk group serving the Avaya SES proxy (used to reach TFN and any optional SIP endpoints). Note that these parameters also apply to any other elements (such as H.323 phones, MedPro cards, CLANs etc.) also assigned to this region. In the **IP Network Region** form 2:

- The **Authoritative Domain** field is configured to match the domain name configured on the Avaya SES. This field is required for endpoints to call the public network. In this configuration, the domain name ***hw01sx.tfn.net.tw*** is used. Note that this Authorative Domain is set to *dcsip.com* for ip-network-region 1 form as ip-network-region 1 is for enterprise customer site.
- By default, **IP-IP Direct Audio** (shuffling) for both Intra and Inter region, is enabled to allow audio traffic to be sent directly between SIP endpoints without using media resources such as the TN2302AP IP Media Processor (MedPro) card.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this configuration, this codec set will apply to calls with TFN only and does not apply to any IP phone (H.323 or SIP) within the enterprise.
- In page 2, the Source Region 2 to Destination Region 2 **Codec Set** follows the codec set in page 1 of the form. The Source Region 2 to Destination Region 1 **Codec Set** is set as 2 to specify the codec set to be used between SIP Trunk and the enterprise site.

In this case, the SIP trunk is assigned to different IP network region as the G700 Media Gateway. Note also that the **IP Network Region** form is used to set the packet parameters that provides priority treatment for signaling and audio packets over other data traffic on TFN's SIP Trunking service. These parameters may need to be aligned with the specific values provided by TFN.

```
display ip-network-region 2                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 2
Location: 1        Authoritative Domain: hw01sx.tfn.net.tw
    Name: SIP
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3327
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      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: 6
        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
```

```
display ip-network-region 2                                     Page   3 of  19

                   Inter Network Region Connection Management

 src dst codec direct   Total           Video                           Dyn
 rgn rgn  set   WAN  WAN-BW-limits  WAN-BW-limits  Intervening-regions  CAC IGAR
 2   1    2     y        :NoLimit       :NoLimit                              n
 2   2    2
 2   3
 2   4
 2   5
 2   6
 2   7
 2   8
 2   9
 2   10
 2   11
 2   12
 2   13
 2   14
 2   15
```

Figure 7: IP Network Region Form

Step 4: Define IP Codecs

Open the **IP Codec Set** form using the ip-codec value specified in the **IP Network Region** form (**Figure 7**) and enter the audio codec type to be used for calls routed over the SIP trunk. The settings of the **IP Codec Set** form are shown in **Figure 8.** Note that the **IP Codec Set** form may include multiple codecs listed in priority order to allow the codec for the call to be negotiated during call establishment. For TFN, the codecs G.711A, G.711Mu and G.729AB can be supported for incoming. The following is a sample for using G.729AB. During testing, G.729a is not provided for incoming calls to Avaya system. Note that T.38 Fax needs to be set on page 2 of the form to support T.38 Fax calls.

```
change ip-codec-set 2                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.729AB           n          2        20
 2:
 3:
 4:
 5:
 6:
 7:


     Media Encryption
 1: none
 2:
 3:
```

```
change ip-codec-set 2                                          Page   2 of   2

                          IP Codec Set

                              Allow Direct-IP Multimedia? n


                    Mode                 Redundancy
    FAX             t.38-standard         3
    Modem           off                   0
    TDD/TTY         off                   3
    Clear-channel   n                     0
```

Figure 8: IP Codec Set Form

Step 5: Configure the Signaling Groups

For interoperability with TFN, two signaling groups must be configured. One signaling group will be used for outbound calls while the second signaling group will be used for inbound calls. This is necessary because TFN requires that subscribers use a DNS name to reach TFN's proxy server for outbound calls into the TFN network rather than an IP address. This requires the "Far End Domain" field on the signaling group form to be set to the TFN proxy server's DNS name. While this allows outbound Avaya calls through the TFN network to the PSTN, incoming calls will not be able to use this same signaling group because TFN does not use this DNS name when it issues SIP Invite messages. Instead, TFN uses use an IP address. Since Avaya Communication Manager uses the caller's domain/IP address from the SIP Invite message to match up with the "Far End Domain" of a signaling group, there would not be a match. When this happens, Avaya Communication Manager will look for a signaling group with a blank "Far End Domain" field and use this group. If this does not exist, the call will be routed to a random signaling group provided that others exist. In order for inbound calls to then be routed in a deterministic way, another signaling group must be configured with a blank "Far End Domain" field set. This second signaling group can be thought of as a default signaling group. The configuration steps below show how to configure both of these signaling groups.

Configure the *outbound* **Signaling Group** form shown in **Figure 9** as follows:

- Set the **Group Type** field to *sip*.
- The **Transport Method** field will default to *tls* (Transport Layer Security). TLS is the only link protocol that is supported for SIP trunking with Avaya SIP Enablement Services.
- Specify the Avaya S8300B Media server processor (node name "procr") and the Avaya SIP Enablement Services Server (node name "SES") as the two ends of the signaling group in the **Near-end Node Name** and the **Far-end Node Name** fields, respectively. These field values are taken from the **IP Node-Names** form shown in **Figure 6**. For larger media server platforms, the near (local) end of the SIP signaling group may be the CLAN rather than the Avaya S8300B media server processor (procr).
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- Enter the IP Network Region value assigned in the ip-network-region form (**Figure 7**). As the **Far-end Network Region** field is different from the near-end network

region, the preferred codec will be selected from the IP codec set assigned for the inter-regional connectivity for the pair of network regions.

- Enter the domain name of TFN proxy in the **Far-end Domain** field. In this configuration, the domain name is *hw01sx.tfn.net.tw*. This domain is specified in the Uniform Resource Identifier (URI) of the SIP "To" address in the INVITE message. Mis-configuring this field may prevent calls from being successfully established to other SIP endpoints or to the PSTN.
- If calls to/from SIP endpoints are to be shuffled, then the **Direct IP-IP Audio Connections** field must be set to 'y'. In this case, the value will be set to 'y'.
- The **DTMF over IP** field should remain set to the default value of *rtp-payload*. This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833.
- The default values for the other fields may be used.

```
add signaling-group 1
                                SIGNALING GROUP

 Group Number: 1             Group Type: sip
                       Transport Method: tls






   Near-end Node Name: procr                   Far-end Node Name: ses
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region: 2
       Far-end Domain: hw01sx.tfn.net.tw

                                               Bypass If IP Threshold Exceeded? n


         DTMF over IP: rtp-payload              Direct IP-IP Audio Connections? y
                                                          IP Audio Hairpinning? n
 Session Establishment Timer(min): 120
```

Figure 9: Outbound Signaling Group Form

Next, configure the *inbound* **Signaling Group** form following the same steps used for the outbound signaling group above <u>with one exception</u>, leave the **Far-end Domain** field blank as shown in **Figure 10**:

```
add signaling-group 2
                                SIGNALING GROUP

 Group Number: 2             Group Type: sip
                       Transport Method: tls






   Near-end Node Name: procr                   Far-end Node Name: ses
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region: 2
       Far-end Domain:
```

```
                                        Bypass If IP Threshold Exceeded? n

          DTMF over IP: rtp-payload              Direct IP-IP Audio Connections? y
                                                            IP Audio Hairpinning? n
 Session Establishment Timer(min): 120
```

Figure 10: Inbound Signaling Group Form

Step 6: Configure the Trunk Groups
As described above in **Step 5**, two trunks must also be configured. One trunk will be paired with the outbound signaling group and the other with the inbound signaling group.

Configure the *outbound* **Trunk Group** form as shown in **Figure 11** using the "add trunk-group" command. In this case the trunk group number chosen is 1. On Page 1 of this form:

- Set the **Group Type** field to *sip*.
- Choose a mnemonic **Group Name**.
- Specify an available trunk access code (**TAC**).
- Set the **Service Type** field to *tie*.
- Specify the *outbound* signaling group associated with this trunk group in the **Signaling Group** field as previously specified in **Figure 9**.
- Specify the **Number of Members** supported by this SIP trunk group.

Note that one trunk member is required for each call between a non-SIP endpoint and TFN. Calls involving a SIP endpoint and TFN will use two trunk members for the duration of the call.

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

Group Number: 1                        Group Type: sip          CDR Reports: y
  Group Name: SIP TRUNK TO TFN                COR: 995       TN: 1        TAC: #01
   Direction: two-way         Outgoing Display? n
 Dial Access? n                                               Night Service:
Queue Length: 0
Service Type: tie                       Auth Code? n

                                                            Signaling Group: 1
                                                          Number of Members: 10
```

```
add trunk-group 1                                                Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name? y

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 1800
```

Figure 11: Trunk Group Form (Outbound) – Page 1&2

On Page 2 of the **Trunk Group** form:

- set the **Preferred Minimum Session Refresh Interval(sec)** field to the maximum *1800* seconds. This field specifies the refresh INVITE Timer sent to the far-end. One reason that this is adjusted from the default of *120* seconds to *1800* seconds such that outgoing T.38 Fax with G.729b codec will not be dropped. This parameter is adjusted as a workaround solution. Refer to Section 6.2 on the test result and the workaround solution proposed.

On Page 3 of the **Trunk Group** form:

- set the **Numbering Format** field to *public*. This field specifies the format of the calling party number sent to the far-end.

```
add trunk-group 1                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                           Maintenance Tests? y
                     Numbering Format: public
                                              Prepend '+' to Calling Number? n

                                                Replace Unavailable Numbers? n
```

Figure 12: Trunk Group Form (Outbound) – Page 3

Next, configure the *inbound* **Trunk Group** form as shown in **Figure 13** using the “add trunk-group” command. In this case the trunk group number chosen is 2. On Page 1 of this form:

- Set the **Group Type** field to *sip*.
- Choose a mnemonic **Group Name**.
- Specify an available trunk access code (**TAC**).
- Set the **Service Type** field to *tie*.
- Specify the *inbound* signaling group associated with this trunk group in the **Signaling Group** field as previously specified in **Figure 10**.
- Specify the **Number of Members** supported by this SIP trunk group.

Note that one trunk member is required for each call between a non-SIP endpoint and TFN. Calls involving a SIP endpoint and TFN will use two trunk members for the duration of the call.

```
add trunk-group 2                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 2                    Group Type: sip          CDR Reports: y
  Group Name: SIP Trunk Anonymous         COR: 995      TN: 1        TAC: #02
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                          Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                                       Signaling Group: 2
                                                     Number of Members: 10
```

Figure 13: Trunk Group Form (Inbound) – Page 1

On Page 3 of the **Trunk Group** form:

- set the **Numbering Format** field to *public*. This field specifies the format of the calling party number sent to the far-end.

```
add trunk-group 2                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                           Maintenance Tests? y


                     Numbering Format: public
                                              Prepend '+' to Calling Number? n


                                                Replace Unavailable Numbers? n
```

Figure 14: Trunk Group Form (Inbound) – Page 3

Step 7: Configure Calling Party Number Information
Configure the **Numbering Public/Unknown Format** form to send the full calling party number to the far-end.

In this case, all stations with a 5-digit extension beginning with 4 should send the calling party number 0266170054 when an outbound call uses SIP trunk Group #1 (*remember, this the outbound trunk group specified in step 6*). This calling party number will be sent to the far-end in the SIP "From" header. In this case, only one test number was assigned that is used for outgoing calls.

Figure 15 shows the use of the "change public-unknown numbering" command to implement this rule.

```
change public-unknown-numbering 0                                Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                   Total                                   Total
Ext Ext     Trk       CPN           CPN Ext Ext    Trk       CPN            CPN
Len Code    Grp(s)    Prefix        Len Len Code   Grp(s)    Prefix         Len

 5 1        99                       5
 5 4        1         0266170054    10
 5 4        10                       5
 5 4        99                       5
 5 5        99                       5
 5 6        99                       5
```

Figure 15: Numbering Public/Unknown Format Form

Step 8: Automatic Route Selection for Outbound Calls
In these Application Notes, the Automatic Route Selection (ARS) feature will be used to route outbound calls via the SIP trunk to the TFN SIP Trunking Service to a PTSN destination.

Use the **change dialplan analysis** command to add **9** as a feature access code (**fac**).

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                                            Percent Full:    0

      Dialed   Total  Call       Dialed   Total  Call       Dialed   Total  Call
      String   Length Type       String   Length Type       String   Length Type
        1        5    ext
        4        5    ext
        5        5    ext
        6        5    ext
        8        1    fac
        9        1    fac
        *        3    dac
        #        3    dac
```

Figure 16: Change Dialplan Analysis Form

Use the **change feature-access-codes** command to specify **9** as the access code for outside dialing.

```
change feature-access-codes                                     Page   1 of   7
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: *01
                       Answer Back Access Code: *02
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: *03    Deactivation: #03
Call Forwarding Activation Busy/DA: *11    All: *04    Deactivation: #04
                         Call Park Access Code: *05
                       Call Pickup Access Code: *06
CAS Remote Hold/Answer Hold-Unhold Access Code: *07
                  CDR Account Code Access Code: *08
                        Change COR Access Code:
                   Change Coverage Access Code: *09
                   Contact Closure   Open Code:          Close Code:
                   Contact Closure  Pulse Code:
```

Figure 17: Feature Access Codes Form

Next use the **change ars analysis** command to configure the route pattern selection rule based upon the number dialed following the dialed digit "9". In this sample configuration, the PSTN numbers dialed are all in the form AAXXXXXXXX. If the area code (AA) is 02, the call is to be routed to a route pattern containing the SIP trunk groups used for TFN. Note that further administration of ARS is beyond the scope of these Application Notes but discussed in References **[1]** and **[2].**

```
change ars analysis 0                                                    Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    0

          Dialed           Total      Route     Call    Node  ANI
          String          Min  Max   Pattern    Type    Num   Reqd
    02                    10   10     1         pubu          n
```

Figure 18: ARS Analysis Form

Use the **change route-pattern** command to define the SIP trunk group included in the route pattern that ARS selects. In this configuration, route pattern 1 will be used to route calls to trunk group 1, (the SIP trunk created in Step 6**, Figure 11**).

```
change route-pattern 1                                                   Page   1 of   3
                    Pattern Number: 1   Pattern Name: SIP Trunk
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                                 DCS/ IXC
    No          Mrk Lmt List Del  Digits                                   QSIG
                             Dgts                                          Intw
 1: 1    0                                                                  n   user
 2:                                                                         n   user
 3:                                                                         n   user
 4:                                                                         n   user
 5:                                                                         n   user
 6:                                                                         n   user

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

Figure 19: Route Pattern Form

Step 9: Configure Incoming Digit Translation
This step configures the settings necessary to map incoming DID calls to the proper extension(s).

The incoming digits sent in the INVITE message from TFN are manipulated as necessary to route calls to the proper extension on Avaya Communication Manager. Note that this step cannot be completed until the DID numbers and routing strategy defined in Sections 4.1 and 5 are known. Return to this step after the Section 5 work is completed if necessary.

In the examples used in these Application Notes, the incoming DID numbers provided by TFN do not have a direct correlation to the internal extensions assigned within Avaya Communication Manager. Thus all incoming called number digits are deleted and replaced by the assigned extension number.

To create a fully mapped extension number as shown in **Figure 20**:

- Open the **Incoming Call Handling Treatment** form for the *inbound* SIP trunk group configured in **Figure 13**, in this case Trunk Group 2.
- For each extension assigned a DID number from TFN, enter **10** into the **Called Len** and **Del** fields, and the entire **10 digit DID number** into the **Called Number** field.
  Enter the desired Avaya Communication Manager extension number into the **Insert** field.

```
change inc-call-handling-trmt trunk-group 2                    Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/       Called    Called        Del  Insert
 Feature         Len       Number
 tie              10  0266170054        10   40001
 tie
 tie
 tie
 tie
 tie
 tie
 tie
 tie
 tie
 tie
 tie
 tie
 tie
 tie
 tie
 tie
```

Figure 20: Incoming Call Handling Treatment – Full Extension Mapping

If the customer’s extension numbering aligns with the DID numbers (i.e., the final DID digits match the extension), it is not necessary to define an entry for each DID number. Assuming a PBX dial plan that used the 5 digit extensions 40000 thru 49999 and assuming TFN provided DID numbers of 02-6617-0000 thru 0099, the incoming number translation would be done similar to **Figure 21**. Note that the Called Number entry in this case represents the common matching portion applicable to all incoming numbers. Thus 02661700 matches all numbers in the assigned DID block from TFN.

```
change inc-call-handling-trmt trunk-group 2                     Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/       Called    Called        Del  Insert
 Feature         Len       Number
 tie              10  02661700            10   40000
```

Figure 21: Incoming Call Handling Treatment – Simple Extension Mapping

Step 10: Save Avaya Communication Manager Changes
Enter “save translation” to make the changes permanent.

3.2 SIP Endpoint Configuration

This section describes the administration of SIP telephones and requires the preceding SIP Trunk configuration to have been completed. SIP telephones are optional and not required to use the TFN SIP Trunking Service.

Step 1: **Assign a Station**
The first step in adding an off-PBX station (OPS) for Avaya SIP telephones registered with Avaya SIP Enablement Services is to assign a station as shown in **Figure 22**.
Using the “**add station**” command from the SAT:

- Leave the station **Type** at the default “6408D+” value. (Note this is the Avaya recommended best practice that will prevent an alarm warning that occurs when 4600 series phone models are entered).
- Enter “X” in the **Port** field to indicate station administration without port hardware.
- Enter a **Name** for the station that will be displayed.
- The **Security Code** is left blank for SIP OPS extensions.

The remaining fields are configured per normal station administration that is beyond the scope of these Application Notes. Note that the Class of Restrictions (**COR**) and Class of Service (**COS**) will govern the features and call restrictions that apply to this station.

```
add station 40001                                            Page   1 of   4
                                     STATION

Extension: 40001                          Lock Messages? n          BCC: 0
     Type: 6408D+                         Security Code:            TN: 1
     Port: X                            Coverage Path 1:            COR: 1
     Name: SIP Jenny Lim                Coverage Path 2:            COS: 1
                                        Hunt-to Station:

STATION OPTIONS
              Loss Group: 19            Personalized Ringing Pattern: 1
                                                    Message Lamp Ext: 40001
            Speakerphone: 2-way                  Mute Button Enabled? y
        Display Language: english                   Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal                 Media Complex Ext:
   Survivable Trunk Dest? y                             IP SoftPhone? n



                                                 Customizable Labels? y
```

Figure 22: Station Administration – Page 1

On Page 2 of the **Station** form,

- Set the **Restrict Last Appearance** value to 'n' on phones that have 3 or fewer call appearances to maintain proper SIP conference and transfer operation. Setting the **Restrict Last Appearance** value to 'y' reserves the last call appearance for outbound calls. Certain SIP conference and transfer features will not function properly if a third appearance is not available for incoming calls.

```
add station 40001                                            Page   2 of   4
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe              Auto Select Any Idle Appearance? n
          LWC Activation? y                         Coverage Msg Retrieval? y
  LWC Log External Calls? n                                    Auto Answer: none
             CDR Privacy? n                               Data Restriction? n
   Redirect Notification? y                      Idle Appearance Preference? n
 Per Button Ring Control? n                    Bridged Idle Line Preference? n
   Bridged Call Alerting? n                        Restrict Last Appearance? n
  Active Station Ringing: single          Conf/Trans on Primary Appearance? n
                                                         EMU Login Allowed? n
        H.320 Conversion? n           Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
     MWI Served User Type: qsig-mwi                Display Client Redirection? n
                                                  Select Last Used Appearance? n
                                                    Coverage After Forwarding? s

                                               Direct IP-IP Audio Connections? y
 Emergency Location Ext: 40001     Always Use? n       IP Audio Hairpinning? n
```

Figure 23: Station Administration – Page 2

On Page 3 of the **Station** form, configure 5 call appearances under the **Button Assignments** section for the SIP telephone, as shown in **Figure 24**.

```
add station 40001                                            Page   3 of   4
                                     STATION
 SITE DATA
       Room:                                            Headset? n
       Jack:                                            Speaker? n
      Cable:                                           Mounting: d
      Floor:                                       Cord Length: 0
   Building:                                          Set Color:



ABBREVIATED DIALING
     List1:                   List2:                   List3:

BUTTON ASSIGNMENTS
 1: call-appr                           5: call-appr
 2: call-appr                           6:
 3: call-appr                           7:
 4: call-appr                           8:
```

Figure 24: Station Administration – Page 3

A similar number of call appearances should be configured on the SIP Telephone which is beyond the scope of these Application Notes. The parameters to administer call appearances (and many other settings) are described in Reference **[6]**.

Step 2: **Configure Off-PBX Station Mapping**

The second step of configuring an off-PBX station is to configure the **Off-PBX Telephone** form so that calls destined for a SIP telephone at the enterprise site are routed to Avaya SIP Enablement Services, which will then route the call to the SIP telephone.

On the **Off-PBX-Telephone Station-Mapping** form shown in **Figure 25**:

- Specify the **Station Extension** of the SIP endpoint.
- Set the **Application** field to *OPS*.
- Set the **Phone Number** field to the digits to be sent over the SIP trunk. In this case, the SIP telephone extensions configured on Avaya SIP Enablement Services also match the extensions of the corresponding AWOH stations on Avaya Communication Manager. However, this is not a requirement.
- Set the **Trunk Selection** field to '10', which is the number assigned to the *inbound* SIP trunk group used to route the call to the SIP station. This trunk group number was previously defined for calls to local enterprise SIP stations only.
- Set the **Configuration Set** value. In these Application Notes, Configuration Set 1 uses the default values of the Configuration Set form.

```
change off-pbx-telephone station-mapping 40001                    Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station      Application   Dial   Phone Number      Trunk       Configuration
 Extension                  Prefix                   Selection   Set
 40001          OPS             - 40001               10         1
```

Figure 25: Stations with Off-PBX Telephone Integration – Page 1

On Page 2, set the **Call Limit** field to the maximum number of calls that may be active simultaneously at the station. In this example, the call limit is set to '5', which corresponds to the number of call appearances configured on the station form. Accept the default values for the other fields.

```
change off-pbx-telephone station-mapping 40001                    Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Call       Mapping      Calls       Bridged
 Extension      Limit      Mode         Allowed     Calls
 40001         5         both          all         both
```

Figure 26: Stations with Off-PBX Telephone Integration – Page 2

Step 3: Repeat for each SIP Phone

Repeat Steps 1 and 2 for each SIP phone to be added.

Step 4: Save Avaya Communication Manager Changes
Enter "save translation" to make the changes permanent.

4. Configure Avaya SIP Enablement Services

This section covers the administration of Avaya SIP Enablement Services (SES). Avaya SIP Enablement Services is configured via an Internet browser using the Administration web interface. It is assumed that Avaya SIP Enablement Services software and the license file have already been installed on Avaya SIP Enablement Services. During the software installation, the `initial_setup` script is run on the Linux shell of the server to specify the IP network properties of the server along with other parameters. For additional information on these installation tasks, refer to **[4]**.

This section is divided into two parts: Section 4.1 provides the steps necessary to configure SIP trunking to TFN's Global SIP Trunking Service. Section 4.2 provides the steps necessary to complete the administration for optional SIP endpoints (whose configuration was begun on Avaya Communication Manager in Section 3.2).

4.1. SIP Trunking to TFN

Step 1: Access Avaya SIP Enablement Services

Access the SES Administration web interface, by entering http://<ip-addr>/admin as the URL in an Internet browser, where *<ip-addr>* is the IP address of Avaya SIP Enablement Services server.

Log in with the appropriate credentials and then select the *Launch Administration Web Interface* link from the main screen as shown in **Figure 27**.

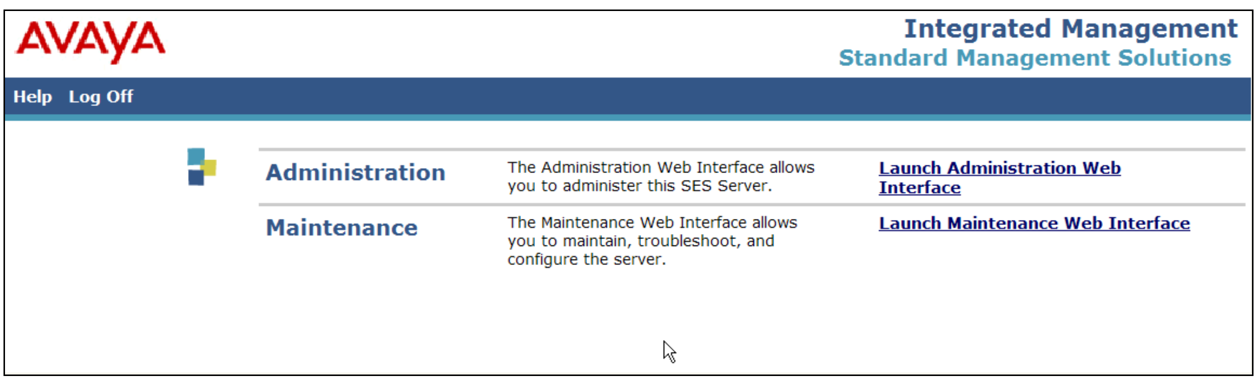

Figure 27 - Avaya SES Main Screen

The SES administration home screen shown in **Figure 28** should be displayed.

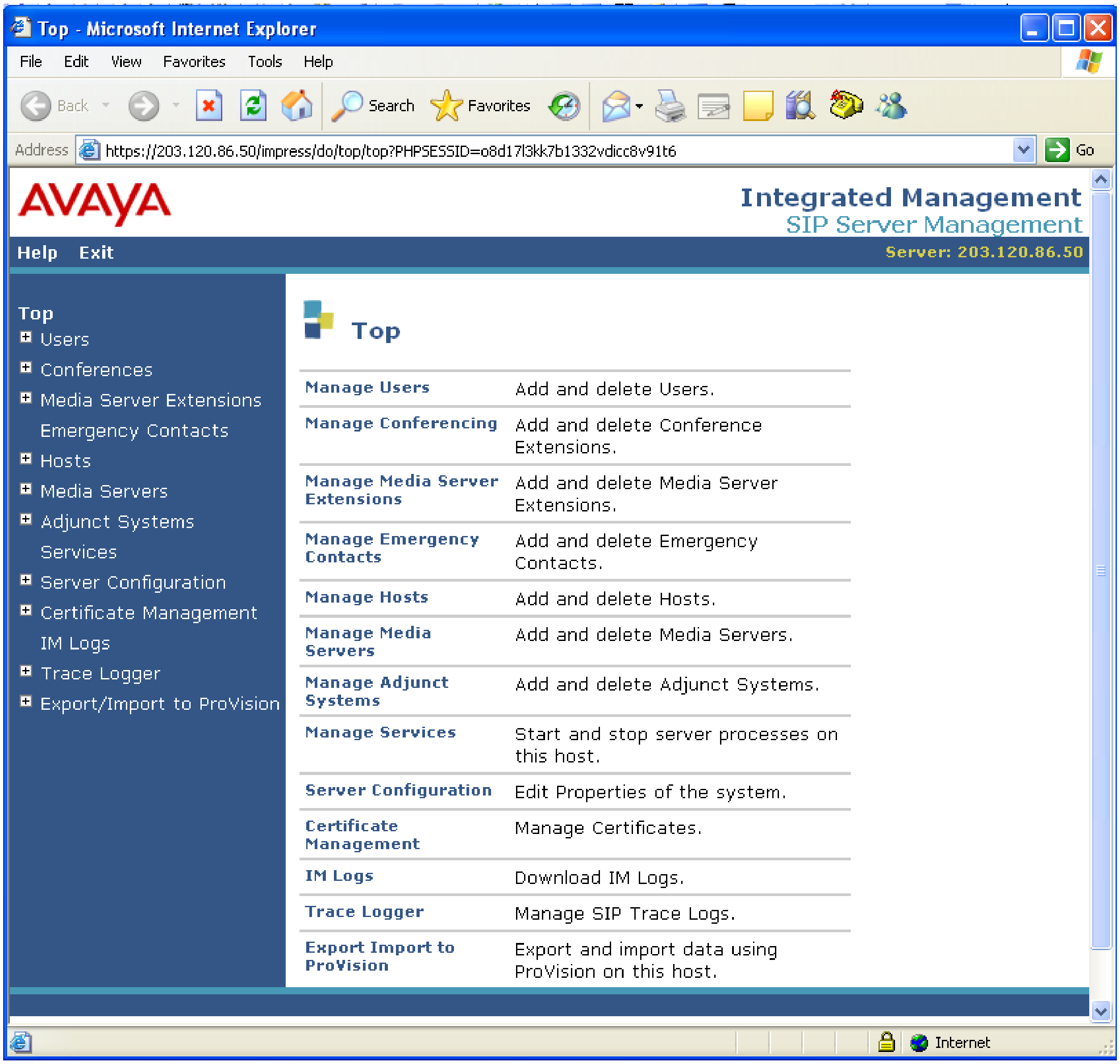

Figure 28: Avaya SES Administration Home Page

Step 2: Define System Properties

From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. This screen displays the Avaya SES version and the network properties entered via the `initial_setup` script during the installation process.

In the **System Properties** screen,

- Enter the **SIP Domain** name assigned to Avaya SES. In this configuration, the SIP domain is "dcsip.com".
- Enter the **License Host** field. This is the host name, the fully qualified domain name, or the IP address of the SIP proxy server that is running the WebLM application and has the associated license file installed. This entry should always be **localhost** unless the WebLM server is not co-resident with this server.
- After configuring the **System Properties** screen, shown in **Figure 29**, click the **Update** button.

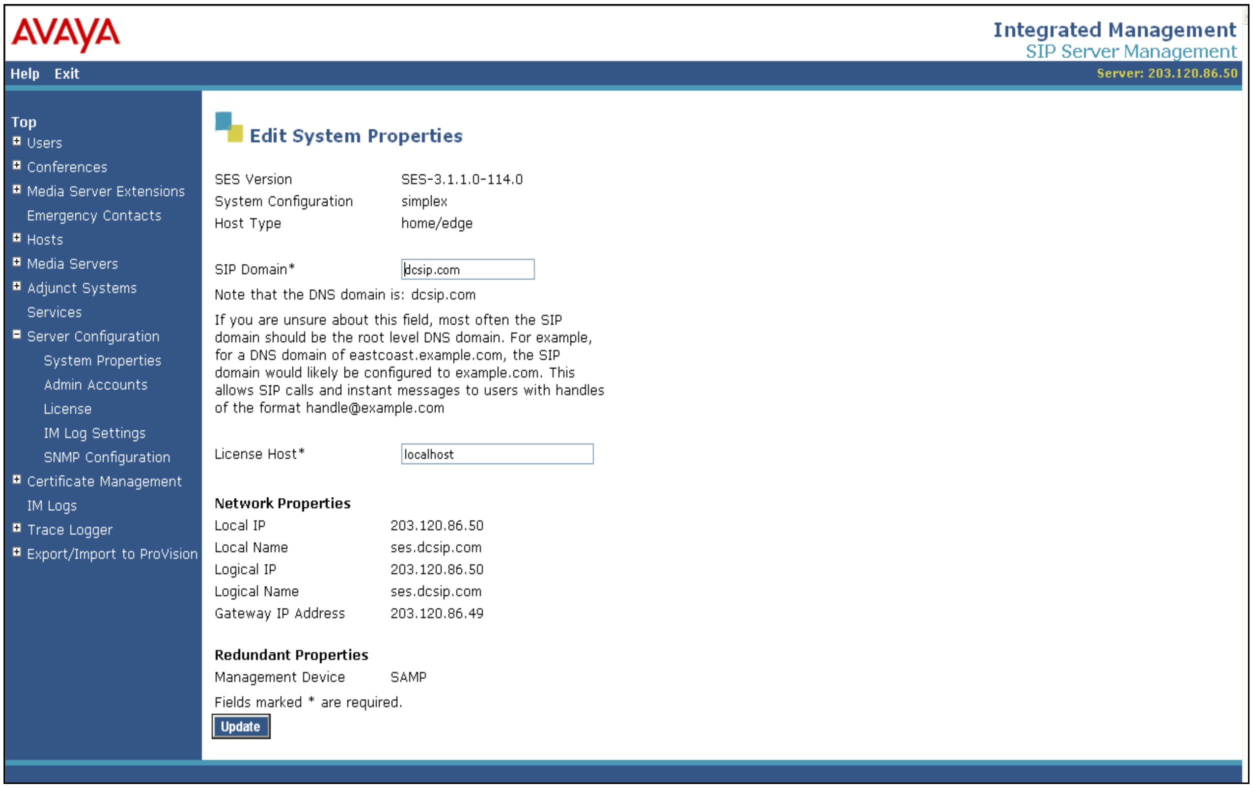

Figure 29: System Properties

Step 3: Enter Avaya SES Host Information

After setting up the domain in the **System Properties** screen, create a host computer entry for Avaya SIP Enablement Services. The following example shows the **Edit Host** screen since the host had already been added to the system.

The **Edit Host** screen shown in **Figure 30** is accessible by clicking on the **Hosts** link in the left pane and then clicking on the **edit** option under the **Commands** section of the subsequent page that is displayed.

- Enter the **Logical IP** or **Logical Name** (shown in **Figure 30**) of this server in the **Host IP Address** field.
- Enter the **DB Password** that was specified while running the `initial_setup` script during the system installation.
- The default values for the other fields may be used as shown in **Figure 30**.
- Click the **Update** button.

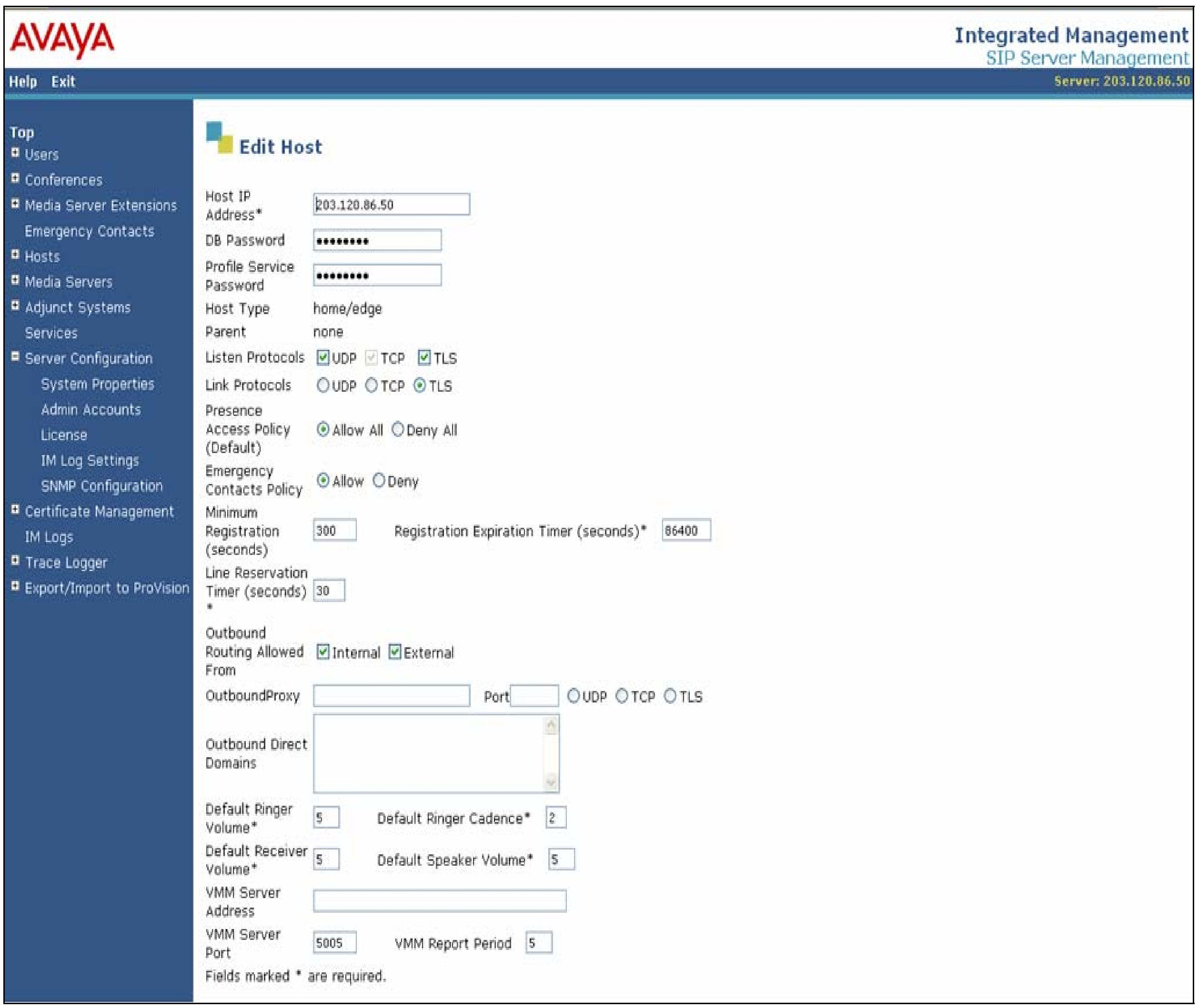

Figure 30: Edit Host

Step 4: **Add Avaya Communication Manager as Media Server**

Under the **Media Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server in the enterprise site. This will create the Avaya SES side of the SIP trunk previously created in Avaya Communication Manager.

In the **Add Media Server** screen, enter the following information:

- A descriptive name in the **Media Server Interface** field (e.g., procr).
- Select the home SES server in the **Host** field as specified in **Figure 30**.

  Select *TLS* (Transport Link Security) for the **Link Type**. TLS provides encryption at the transport layer.
- Enter the IP address of the Avaya S8300B Media Server processor in the **SIP Trunk IP Address** field. (Note: This may be the IP address of the CLAN board in larger Avaya Communication Manager configurations such as an Avaya S8720 Media Server using an Avaya G650 Media Gateway.)
- After completing the **Add Media Server** screen, click on the **Add** button.

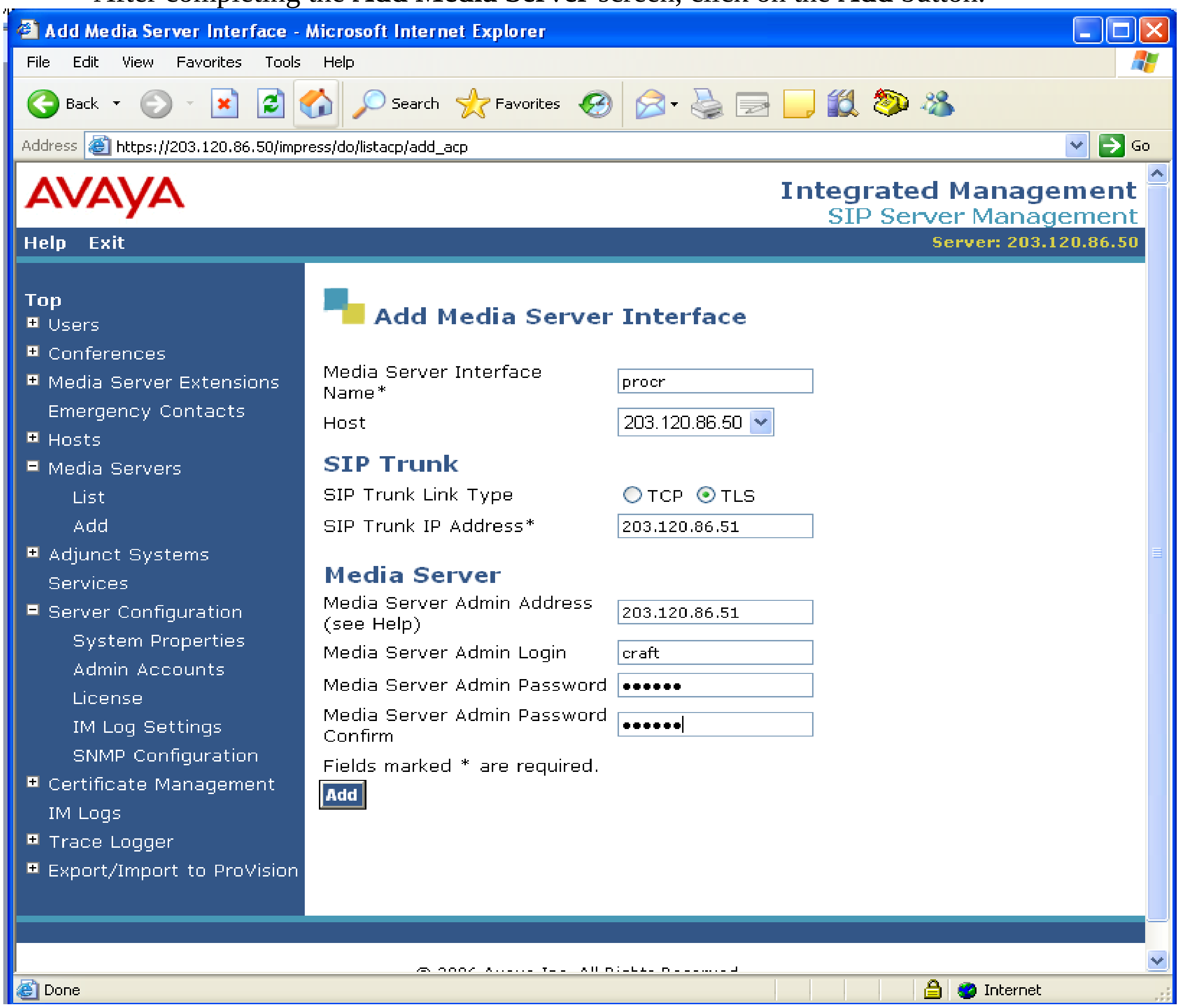

Figure 31: Add Media Server

Step 5: Specify Address Maps to Media Servers

Incoming calls arriving at Avaya SES are routed to the appropriate Avaya Communication Manager for termination services. This routing is specified in a Media Server Address Map configured on Avaya SES.

This routing compares the Uniform Resource Identifier (URI) of an incoming INVITE message to the pattern configured in the Media Server Address Map**,** and if there is a match, the call is routed to the designated Avaya Communication Manager. The URI usually takes the form of *sip:user@domain*, where *domain* can be a domain name or an IP address. Patterns must be specific enough to uniquely route incoming calls to the proper destination if there are multiple Avaya Communication Manager systems supported by the Avaya SES server.

In these Application Notes, only incoming calls from the PSTN require a media server address map entry. Calls originated by Avaya SIP telephones configured as OPS are automatically routed to the proper Avaya Communication Manager by the assignment of an Avaya Media Server extension to that phone. Address map definitions for SIP endpoints not assigned a media server extension and connections to multiple service providers are beyond the scope of these Application Notes.

For the TFN's SIP Trunking Service, the *user* portion of the SIP URI will contain the 10 digit value specified for the incoming direct inward dialed telephone number. An example of a SIP URI in an INVITE message received from TFN would be:

```
sip:0266170054@203.120.86.50;user=phone
```

The user portion in this case is the 10 digit DID number "0266170054". The strategy used to define the media server address maps will be to create a pattern that matches the DID numbers assigned to the customer by TFN. The Avaya SES will forward the messages with matching patterns to the appropriate media server interface.

To configure a **Media Server Address Map**:

- Select **Media Servers** in the left pane of the Administration web interface. This will display the **List Media Servers** screen.
- Click on the **Map** link associated with the appropriate media server, added in Step 4, to display the **List**
  Media Server Address Map screen.
- Click on the **Add Map In New Group** link. The screen shown in **Figure 32** is displayed. The **Host** field displays the name of the media server that this map applies to.
- Enter a descriptive name in the **Name** field
- Enter the regular expression to be used for the pattern matching in the **Pattern** field.
  In this configuration, the DID numbers provided by TFN are 0266170054. The pattern specification (without the double quotes) for DID numbers assigned is: "^sip:0266170054".
- Click the **Add** button once the form is completed.

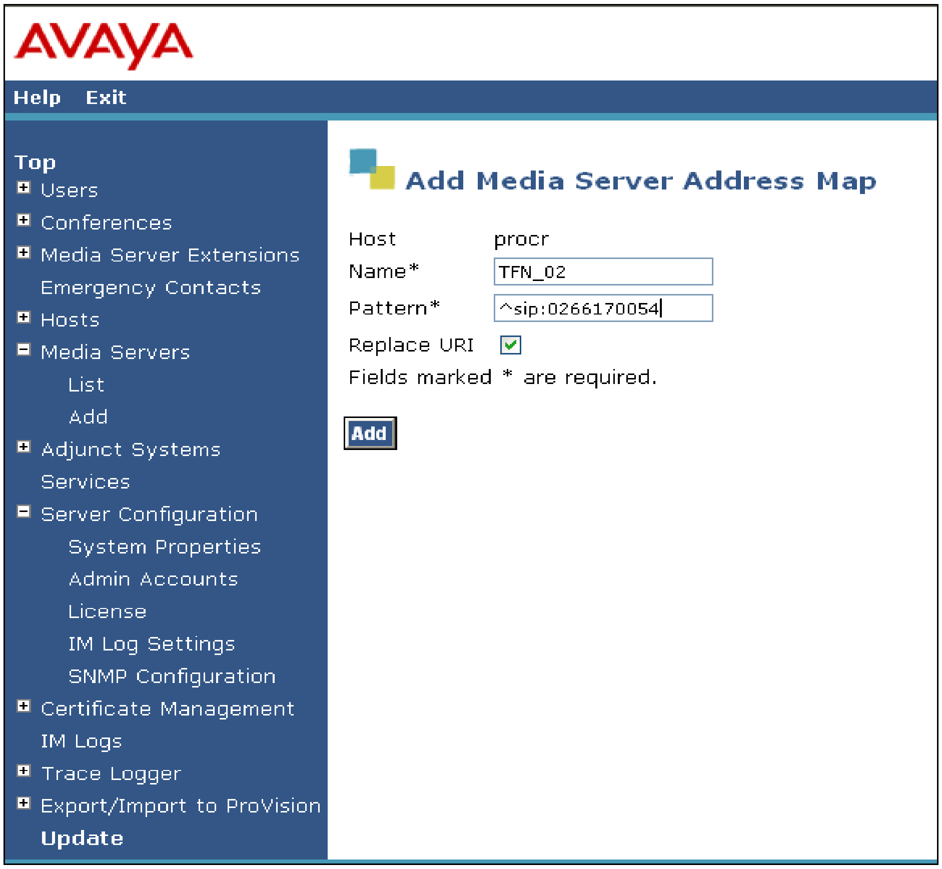

Figure 32: Media Server Address Map

After configuring the media server address map, the **List Media Server Address Map** screen appears as shown in **Figure 33**.

Figure 33: List Media Server Address Map

Note that after the first **Media Server Address Map** is added, the **Media Server Contact** is created automatically. For the **Media Server Address Map** added in **Figure 32**, the following contact was created:

sip:$(user)@203.120.86.51:5061;transport=tls

The contact specifies the processor address of the Avaya S8300B and the transport protocol used to send SIP signaling messages. The incoming DID number sent in the user part of the original request URI is substituted for $(user).

Step 6: Specify Address Maps to TFN

Outbound PSTN calls are directed by Avaya Communication Manager automatic route selection (ARS) according to the customer's network design guidelines. These guidelines determine what types of outgoing calls should be sent to the TFN's SIP Trunking Service. The ARS routing decisions (for trunk group selection) will be customer specific and are beyond the scope of these notes.

SIP signaling messages for outbound calls sent to the SIP trunk are then routed to the TFN gateway using Host Address Maps within Avaya SES. As with the inbound media server address maps, these Host Address Maps use pattern matching on the SIP URI to direct messages to the corresponding contact address (e.g., the TFN SIP signaling gateway). In this configuration, the Avaya SES routing rule for the SIP trunk group will be to send all outbound PSTN traffic to TFN's SIP Trunking Service.

To perform this, several dialing pattern will be created in the Avaya SES.

- The first pattern (without the double quotes) of "`^sip:02817136*`" will match on all sip calls having digits beginning with 02817136.

Note that additional or more specific pattern matches would be used if necessary to selectively route SIP traffic to different destinations (such as multiple service providers serving different geographic regions). Also note that a user dialed access code (such as 9 to place a PSTN call) has been previously deleted (by ARS) prior to seizing the outbound SIP trunk.

The configuration of the host address map for all Taipei local calls is shown in **Figure 30**.

- Access the **Add Host Address Map** screen by selecting the **Hosts** link in the left pane of the Administration web interface and then clicking on the **Map** link associated with the appropriate host. The **List Host Address Map** screen is displayed.
- From this screen, click the **Add Map In New Group** link to display the **Add Host Address Map** screen shown in **Figure 34**. Enter a descriptive name for the map, such as "TFN_Local02".
- Specify an appropriate pattern for the call type. In this example, the pattern used is "^sip:02[0-9]{8}".
- Leave the **Replace URI** checkbox selected.
- Click the **Add** button.

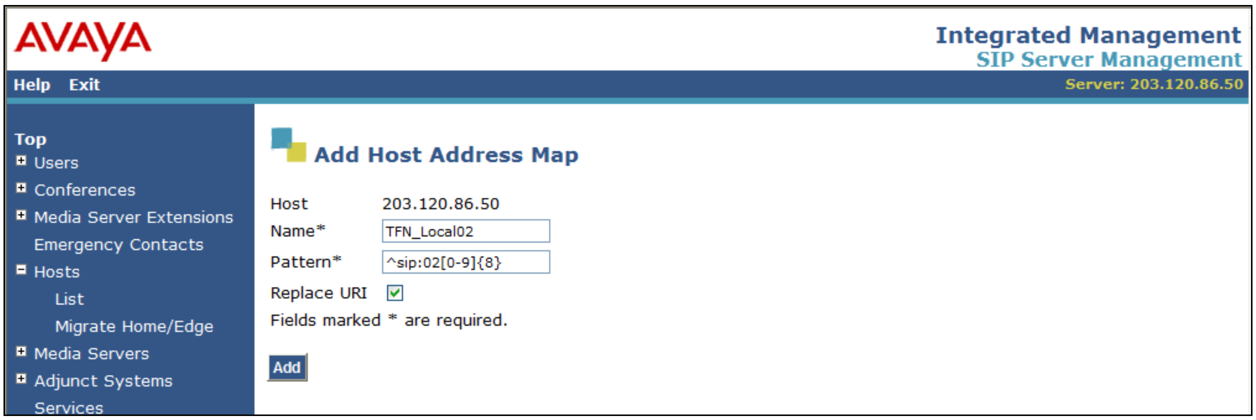

Figure 34: Edit Host Map Entry

Additional Host Address Map patterns are added in a similar manner.

Step 7: Specify the TFN SIP Gateway Information
The next step is to enter the contact address for the TFN SIP gateway. In this example, a DNS name is used to identify TFN's SIP gateway. The customer's specific information will be provided by TFN.

To enter the TFN SIP gateway information:

- As described in Step 6, display the **List Host Address Map** screen.
- Click on the **Add Another Contact** link associated with the address map added in **Figure 34** to open the **Add Host Contact** screen. In this screen, the **Contact** field specifies the destination for the call and it is entered as:

  sip:$(user)@hw01sx.tfn.net.tw

  The user part in the original request URI is inserted in place of the "$(user)" string before the message is sent to TFN.
- Click the **Add** button when completed.

After configuring the host address maps and contact information, the **List Host Address Map** screen will appear as shown in **Figure 35**.

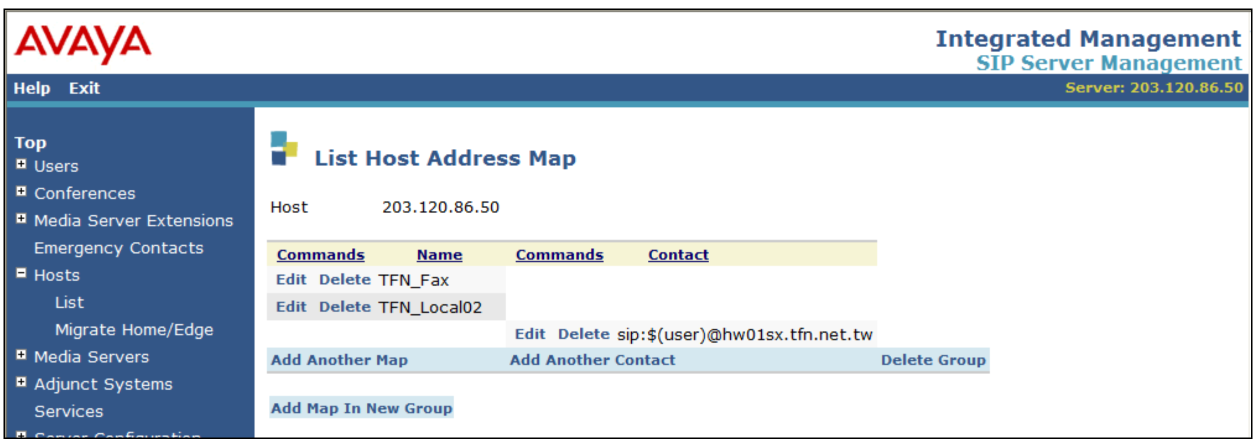

Figure 35: List Host Address Map

Step 8: Save the Changes
After making changes within Avaya SES, it is necessary to commit the database changes using the **Update** link that appears when changes are pending. Perform this step by clicking on the **Update** link found in the bottom of the blue navigation bar on the left side of any of the Avaya SES Administration screens as shown in **Figure 36**.

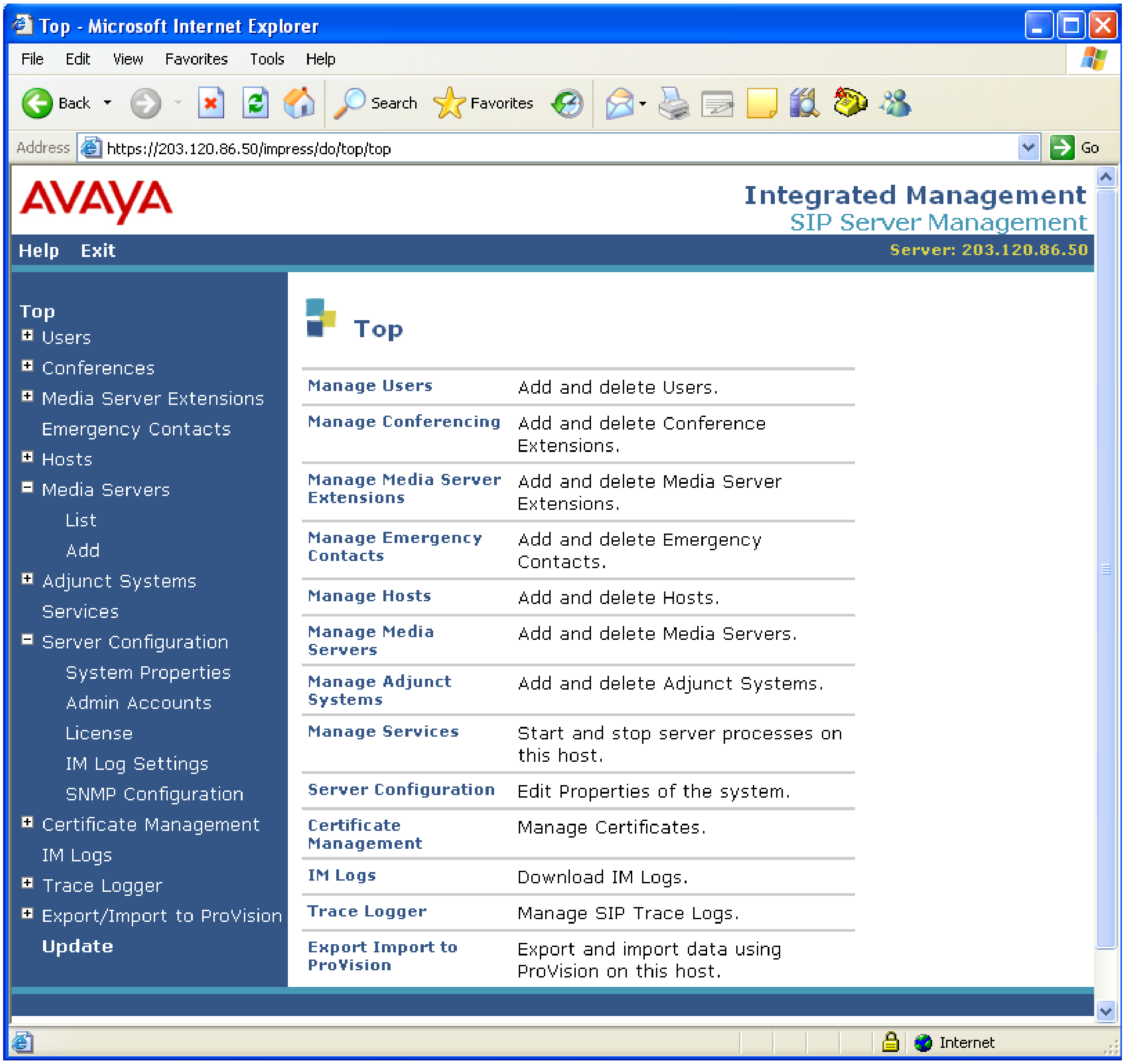

Figure 36: Update Following Avaya SES Administrative Changes

Step 9: Specify the TFN SIP Gateway as a Trusted Host
The final step to complete the SIP trunk administration on Avaya SES is to designate the IP address of TFN SIP Gateway as a trusted host. As a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the designated IP address.[1]
If multiple SIP proxies are used, the IP address of each SIP proxy must be added as a trusted host.

To configure a trusted host:

- Log in to the Avaya SES via the Linux server, using the administrative login and password.
- Enter the following trustedhost command at the Linux shell prompt:
  trustedhost –a 61.30.231.130 –n 203.120.86.50 –c TFN
  The –a argument specifies the address to be trusted; –n specifies the Avaya SES host name; –c adds a comment.
- Use the following trustedhost command to verify the entry is correct:
  trustedhost –L

Figure 37 illustrates the results of the trustedhost commands.[2]

- Complete the trusted host configuration by returning to the main Avaya SES Administration web page and again clicking on the **Update** link as shown in **Figure 36**.

  If the **Update** link is not visible, refresh the page by selecting **Top** from the left hand menu. Note this step is required even though the trusted host was configured via the Linux shell.

```
craft@ses> trustedhost -a 61.30.231.130 -n 203.120.86.50 -c TFN
61.30.231.130 is added to trusted host list.
craft@ses> trustedhost -L
Third party trusted hosts.
 Trusted Host IP address  |   SES Host IP address     |         Comment
--------------------------+---------------------------+-----------------------
61.30.231.130             | 203.120.86.50             | TFN

craft@ses>
```

Figure 37: Configuring a Trusted Host

---

[1] Note, if the trusted host step is not done, authentication challenges to incoming SIP messages (such as INVITEs and BYEs) will be issued by the SES. This may cause call setup to fail, active calls to be disconnected after timeout periods, and/or SIP protocol errors.

[2] For completeness, the –d argument allows the trust relationship to be deleted. For, example,

trustedhost –d proxyX.TFN –n 10.1.1.124

removes the trust relationship.

4.2. Configuration for SIP Telephones

This section provides basic instructions for completing the administration necessary to support the optional Avaya 46xx SIP telephones. Additional features such as the use of mnemonic addressing and instant messaging are also supported by Avaya SES but are beyond the scope of these Application Notes.

Step 1: Add a SIP User

Create the SIP user record as follows:

- In Avaya SES administration, expand the **Users** link in the left side blue navigation bar and click on the **Add** link.
- In the **Add User** screen, enter the extension of the SIP endpoint in the **Primary Handle** field.
- Enter a user password in the **Password** and **Confirm Password** fields. This password will be used when logging into the user's SIP telephone.
- In the **Host** field, select the Avaya SES server hosting the domain (203.120.86.50) for this user. Enter the **First Name** and **Last Name** of the user.
- To associate a media server extension with this user, select the **Add Media Server Extension** checkbox. Calls from this user will always be routed through Avaya Communication Manager over the SIP trunk for origination services.
- Press the **Add** button. This will cause a confirmation screen to appear.
- Press **Continue** on the confirmation screen.

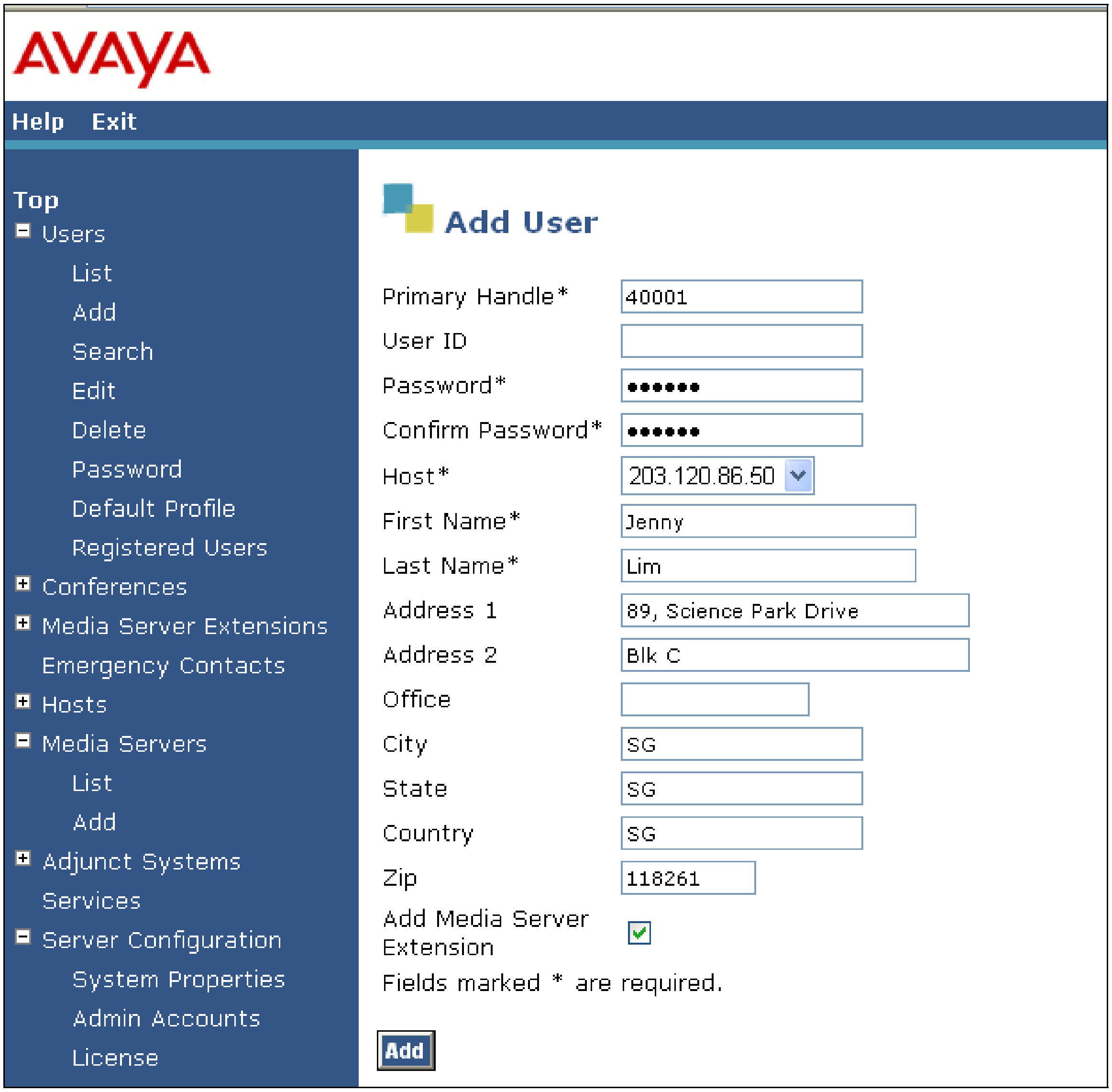

Figure 38: Add User

Step 2: Specify Corresponding Avaya Communication Manager Extension
The SIP phone handle must now be associated with the corresponding extension on Avaya Communication Manager.

- In the **Add Media Server Extension** screen, enter the **Extension** configured on the media server, shown in **Figure 22**, for the OPS extension on Avaya Communication Manager previously defined in Section 3.2. Usually, the media server extension and the user extension are the same (recommended) but it is not required to be.
- Select the **Media Server** assigned to this extension.
- Click on the **Add** button.

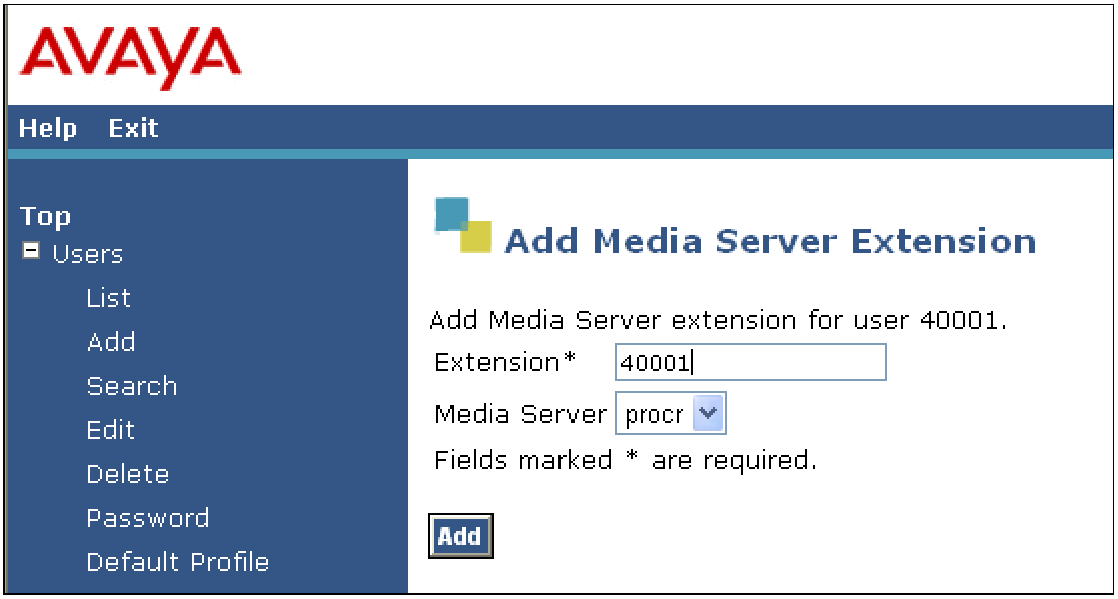

Figure 39: Add Media Server Extension

Step 3: Repeat for Each SIP User
Repeat Steps 1 and 2 for each SIP user.

5. TFN SIP Trunking Services Configuration

In order to use TFN VoIP Services, a customer must order service from TFN using the TFN sales processes. The process can be started by contacting TFN via their corporate web site at http://www.tfn.net.tw or by contacting a TFN Business sales representative through this number **0809 000188.**

6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between TFN's SIP Trunking Service and an Avaya IP Telephony Solution. This section covers the general test approach and the test results.

6.1. General Test Approach

A simulated enterprise site consisting of an Avaya IP telephony solution supporting SIP trunking was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the commercially available SIP Trunking Service provided by TFN. This allowed the enterprise site to use SIP trunking for PSTN calling.

The following features and functionality were covered during the SIP trunking interoperability compliance test:

- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by TFN.
- Outgoing calls from the enterprise site were completed via TFN to the PSTN destinations.
- Calls using SIP, H.323, digital and analog endpoints supported by the Avaya IP telephony solution.
- Calls using G.711 codec and G.729.
- Fax routing to ensure G.711 and G.729 use for fax calls.
- DTMF tone transmission using RFC 2833 with successful Voice Mail /Vector navigation.
- Telephone features such as hold, transfer, conference.
- Direct IP-to-IP media (also known as "shuffling") with SIP/H323 telephones.

6.2. Test Results

Interoperability testing of the sample configuration was completed with successful results.

The following items described in **Table 2** below were observed.

Item	Issue Observed	Discussion / Workaround
Avaya Communication Manager Codec behavior.	Using a codec list of G.729 followed by G711Mu can cause calls from SIP Telephones to disconnect after shuffling or using the HOLD feature.	This can be avoided by not using this combination/order in Avaya Communication Manager codec list.
T.38 Fax Outgoing with Multiple pages.	An outbound fax with multiple pages may result in a dropped connection with a communication error.	A workaround for this issue is to extend the refresh timer on the SIP Trunk to maximum of 1800 seconds.
Conference with Incoming call to Avaya SIP telephone.	Incoming PSTN call to Avaya SIP phone and conferencing to another internal Avaya Communication Manager extension causes the PSTN caller and the rest of the party to have no talk path.	This is caused by misaligned Telephone events. Avaya accepts whatever Telephone event for incoming calls. TFN is using Telephone event 97 and the Avaya 4600 series SIP Telephone uses 127. This mismatch in Telephone event causes an audio issue when shuffling is turn on. Until this interoperability issue is resolved, a workaround is to place the Avaya SIP Telephones in a separate IP Network-region and disable the inter-region shuffling.

Table 2: Interoperability Observations

7. Verification Steps

This section provides verification steps that may be performed in the field to verify that the SIP, H.323, digital and analog endpoints can place outbound and receive inbound PSTN calls through TFN.

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.

2. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.

3. Verify that the user on the PSTN can terminate an active call by hanging up.

4. Verify that an endpoint at the enterprise site can terminate an active call by hanging up.

8. Support

For technical support on TFN's SIP Trunking Service, contact TFN Customer Service at local number ***0809 000188***

9. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya Communication Manager and Avaya SES telephony solution to TFN's SIP Trunking Service. SIP trunking uses the Session Initiation Protocol (SIP) to connect private company networks to the public telephone network via converged IP access. It provides businesses a flexible, cost-saving alternative to traditional hardwired telephony trunk lines.

10. References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at http://support.avaya.com.

[1] *Administrator Guide for Avaya Communication Manager*, May 2006, Issue 2.1, Document Number 03-300509.
[2] *Feature Description and Implementation for Avaya Communication Manager,* February 2006, Issue 4, Document Number 555-245-205
[3] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.1*, February 2006, Issue 9, Document Number 210-100-700.
[4] *Installing and Administering SIP Enablement Services R3.1.1,* August 2006, Issue 2.0, Document Number 03-600768
[5] *SIP Support in Release 3.1 of Avaya Communication Manager Running on the Avaya S8300,S8500, S8500B, S8700, and S8710 Media Server*, February 2006, Issue 6, Document Number 555-245-206.
[6] *4600 Series IP Telephone LAN Administrator Guide,* February 2007, Issue 6, Document Number 555-233-507

APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by TFN and the Avaya SIP network at the enterprise site. Customers may use these INVITE messages for comparison and troubleshooting purposes. Differences in these messages may indicate different configuration options selected.

Sample SIP INVITE Message from TFN to Avaya SIP Enablement Services:

```
Session Initiation Protocol
Request-Line: INVITE sip:0266170054@203.120.86.50;user=phone SIP/2.0
Message Header
        Via: SIP/2.0/UDP 61.30.231.130:5062;branch=z9hG4bK28ff96a9a
        Call-ID: 6c1cd91d3ab72e1abf1d5446545e7a04@61.30.231.130
        From: <sip:0281713600@61.30.231.130;user=phone>;tag=89ea08b7
        To: <sip:0266170054@203.120.86.50;user=phone>
        CSeq: 1 INVITE
        Contact: <sip:0281713600@61.30.231.130:5062;user=phone>
        Supported: 100rel
        User-Agent: Huawei SoftX3000 R006B03D
        Max-Forwards: 70
        Allow:
INVITE,ACK,CANCEL,OPTIONS,BYE,REGISTER,PRACK,INFO,UPDATE,SUBSCRIBE,NOTIFY,MESSAGE,REFER
        Content-Length: 337
        Content-Type: application/sdp
Message body
Session Description Protocol
        Session Description Protocol Version (v): 0
        Owner/Creator, Session Id (o): HuaweiSoftX3000 655001 655001 IN IP4 61.30.231.130
        Session Name (s): Sip Call
        Connection Information (c): IN IP4 61.30.231.250
        Time Description, active time (t): 0 0
        Media Description, name and address (m): audio 29596 RTP/AVP 18 8 0 4 2 97
        Media Attribute (a): rtpmap:18 G729/8000
        Media Attribute (a): rtpmap:8 PCMA/8000
        Media Attribute (a): rtpmap:0 PCMU/8000
        Media Attribute (a): rtpmap:4 G723/8000
        Media Attribute (a): rtpmap:2 G726-32/8000
        Media Attribute (a): rtpmap:97 telephone-event/8000
        Media Attribute (a): ptime:20
        Media Attribute (a): fmtp:97 0-15
        Media Attribute (a): fmtp:18 annexb=yes
```

Sample SIP INVITE Message from Avaya SIP Enablement Services to TFN:

```
Session Initiation Protocol
Request-Line: INVITE sip:0281713608@hw01sx.tfn.net.tw SIP/2.0
Message Header
        Call-ID: 0f01c5a4ddcdb1227146eb0500
        CSeq: 1 INVITE
        From: "SIP Jenny Lim" <sip:anonymous.invalid:5061>;tag=0f01c5a4ddcdb1217146eb0500
        Record-Route: <sip:203.120.86.50:5060;lr>,<sip:203.120.86.51:5061;lr;transport=tls>
        To: "0281713608" <sip:0281713608@hw01sx.tfn.net.tw>
        Via: SIP/2.0/UDP 203.120.86.50:5060;branch=z9hG4bK030303666666030303d3ef.0,SIP/2.0/TLS
203.120.86.51;Content-Length: 156
        Content-Type: application/sdp
        Contact: "SIP Jenny Lim" <sip:203.120.86.51:5061;transport=tls>
        Max-Forwards: 67
        User-Agent: Avaya CM/R013x.01.2.632.1
        Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
        History-Info: <sip:0281713608@hw01sx.tfn.net.tw>;index=1
        History-Info: "0281713608" <sip:0281713608@hw01sx.tfn.net.tw>;index=1.1
        Supported: 100rel,timer,replaces,join,histinfo
        Min-SE: 240
        Session-Expires: 240;refresher=uac
        Privacy: id
Message body
Session Description Protocol
   Session Description Protocol Version (v): 0
        Owner/Creator, Session Id (o): - 1 1 IN IP4 203.120.86.51
        Session Name (s): -
        Connection Information (c): IN IP4 203.120.86.54
        Time Description, active time (t): 0 0
        Media Description, name and address (m): audio 2054 RTP/AVP 0 127
        Media Attribute (a): rtpmap:0 PCMU/8000
        Media Attribute (a): rtpmap:127 telephone-event/8000
```

APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within the Avaya SES is a Linux regular expression used to match against the URI string found in the SIP INVITE message.
Regular expressions are a way to describe text through pattern matching. The regular expression is a string containing a combination of normal text characters, which match themselves, and special *metacharacters*, which may represent items like quantity, location or types of character(s).

In the pattern matching string used in the Avaya SES:

- Normal text characters and numbers match themselves.
- Common metacharacters used are:
  - A period **.** matches any character once (and only once).
  - An asterisk ***** matches zero or more of the preceding characters.
  - Square brackets enclose a list of any character to be matched. Ranges are designated by using a hyphen. Thus the expression **[12345]** or **[1-5]** both describe a pattern that will match any single digit between 1 and 5.
  - Curly brackets containing an integer 'n' indicate that the preceding character must be matched exactly 'n' times. Thus **5{3}** matches '555' and **[0-9]{10}** indicates any 10 digit number.
  - The circumflex character ^ as the first character in the pattern indicates that the string must begin with the character following the circumflex.
    Putting these constructs together as used in this document, the pattern to match the SIP INVITE string for any valid 1+ 10 digit number in the North American dial plan would be:
    ^sip:1[0-9]{10}

    This reads as: "Strings that begin with exactly **sip:1** and having any 10 digits following will match.

A typical INVITE request below uses the shaded portion to illustrate the matching pattern.

```
INVITE sip:0266170054@203.120.86.50;user=phone SIP/2.0
```

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.