Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between SingTel SIP Trunking Service on the ConnectPlus IP Network and an Avaya IP Telephony Solution – Issue 1.0

Abstract

These Application Notes describe the steps to configure SIP Trunking between SingTel SIP Trunking service on the ConnectPlus IP Network and an Avaya IP Telephony solution. The Avaya solution consists of Avaya SIP Enablement Services, Avaya Communication Manager, and various Avaya SIP, H.323 and analog end points.

SingTel is one of the largest service providers in Singapore and the region. The ConnectPlus is a fully managed private IP Network that spans a number of countries in Asia Pacific through a consortium of telecommunications carriers. Certification of SingTel's SIP trunking service will allow new and existing customers on this network to connect using SIP trunks with Avaya PBXs.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program onsite at SingTel ComCentre.

1. Introduction

These Application Notes describe the steps to configure SIP trunking between the SingTel SIP Trunking Service on the ConnectPlus IP Network and an Avaya IP telephony solution. The Avaya solution consists of Avaya SIP Enablement Services, Avaya Communication Manager and various Avaya telephony endpoints. These endpoints include IP telephones (using SIP and H.323 protocols) and traditional analog phones.

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1.1 SIP Trunking Configuration

Figure 1 illustrates a sample Avaya IP telephony solution connected to SingTel's SIP Trunking service. This is the configuration used during the DevConnect compliance testing process. Note that a simulated Layer 2 IP network was used in the setup instead of using MPLS network.

The Avaya IP telephony solution used to create a simulated customer site contained:
• An Avaya S8300B Server with an Avaya G350 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.
• Avaya 4600 series IP telephones (configured to use either the SIP or H.323 protocol), Avaya One-X series IP Telephones (configured to use either the SIP or H.323 protocol), Avaya Digital (DCP) telephones or analog telephones.

In this configuration, two terms are being used for the Singtel Network are On-Net and Off-Net. On-Net are defined as calls within the IP Network whereas Off-Net are defined as calls out of the IP Network via the PSTN Gateway. There is no provision for inbound calls from the PSTN to users in the IP Network.

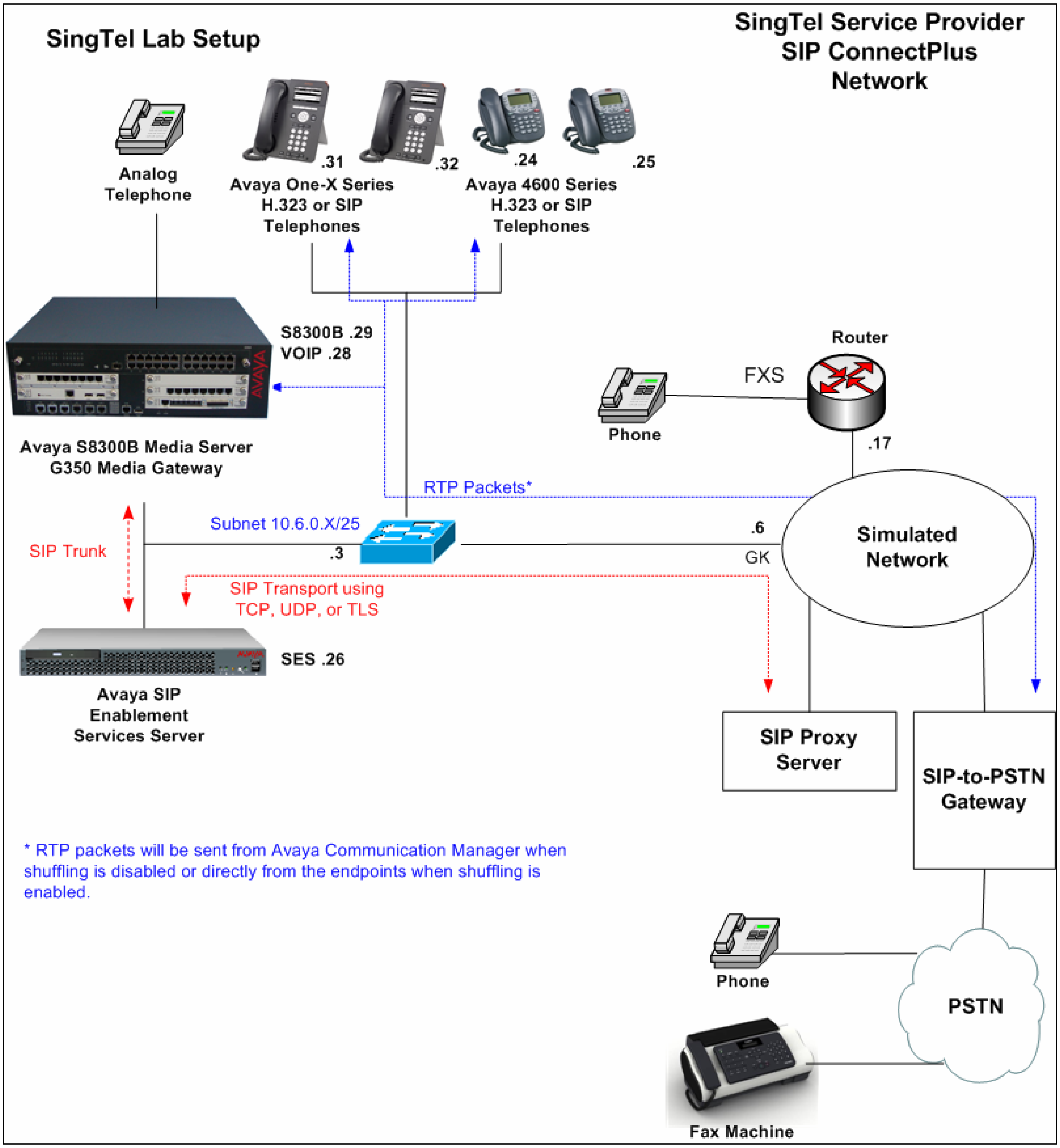

Figure 1: Avaya IP Telephony Network using SingTel ConnectPlus SIP Trunking Service

1.2 Call Flows

To better understand how calls are routed between the users on the IP Network 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 an On-Net call to the enterprise site terminating on a typical analog telephone supported by Avaya Communication Manager.

1. An On-Net user dials a SingTel provided DID number assigned to an Avaya Communication Manager telephone at the enterprise site. The Gatekeeper routes the call to the SingTel network (as the local service provider) which routes the DID number to the assigned customer.

2. Based on the DID number, SingTel 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 or 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.)

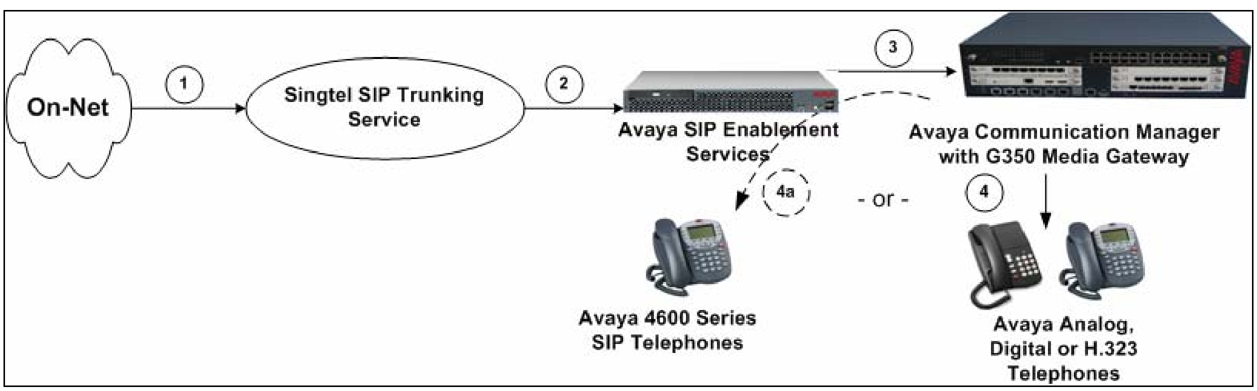

Figure 2: Incoming On-Net Calls to Avaya Communication Manager

Appendix A illustrates an example of a SIP INVITE message sent by SingTel 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 or On-Net via the SIP trunk to SingTel.

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 SingTel.

4. SingTel completes the call to the PSTN or On-Net.

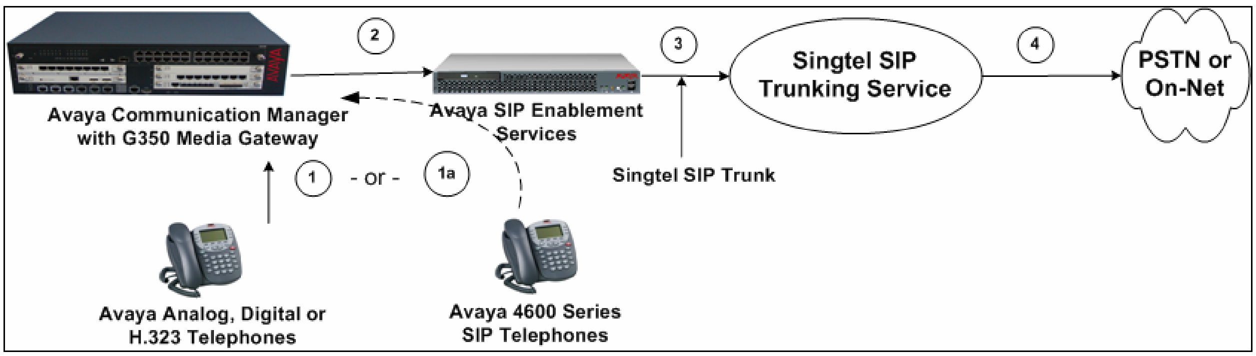

Figure 3: Outgoing Calls from Avaya Communication Manager to the PSTN or On-Net

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 4.0.1 Service Pack 14300 (R014x.00.1.731.2)
Avaya G350 Media Gateway	MGP: 26.33.0 VOIP: 70 Onboard Analog: HW6/FW83
Avaya SIP Enablement Services R4.0 on S8500B Media Server	SES-4.0.0.0-033.6
Avaya 4610SW SIP Telephones	Release 2.2.2
Avaya 4610SW H.323 Telephones	Release 2.8.3
Avaya One-X 9620 H.323 Telephones	Release 1.5
Avaya One-X 9620 SIP Telephones	Release 1.0.13.1
Analog Telephone	n/a
G2 Fax Machine	n/a
SingTel VoIP Service Components	
Component	**Version**
Singtel ConnectPLUS SIP Trunking Service	n/a

Table 1: Equipment and Software Tested

The specific configuration above was used for the SingTel 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 SingTel 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 telephones 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 SingTel. There is no direct SIP signaling path between SingTel 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 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 SingTel gatekeeper.

The dial plan for the configuration described in these Application Notes consists of On-Net 7-digit dialing and Off-Net dialing over the PSTN. However, Directory Assistance calls were not tested as this is not available. 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 Server with G350 Media Gateway is presumed to have been previously completed.

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: V14
       Location: 2                            RFA System ID (SID): 97432
       Platform: 13                           RFA Module ID (MID): 1

                                                            USED
                                Platform Maximum Ports: 900   47
                                      Maximum Stations: 50    6
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 0     0
                    Maximum Off-PBX Telephones -   OPS: 100   2
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 0     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 SingTel network, SIP endpoints and any other SIP trunks used. Note that each SIP OPS telephone on a call with SingTel 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    10
           Maximum Concurrently Registered IP Stations: 50     2
             Maximum Administered Remote Office Trunks: 450    0
Maximum Concurrently Registered Remote Office Stations: 450    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: 450    30

   Maximum Number of DS1 Boards with Echo Cancellation: 80     0
                             Maximum TN2501 VAL Boards: 0      0
                     Maximum Media Gateway VAL Sources: 50     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.)
```

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 “SingTel-SES” and “10.6.0.26” 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   2
                                  IP NODE NAMES
    Name              IP Address
SingTel-SES         10.6.0.26
SingTel-Switch      10.6.0.6
default             0.0.0.0
procr               10.6.0.29









( 5  of 5    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 SingTel 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 6:

- 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 "singtel-sip.com" is used. Note that the Authoritative Domain is set to "*avaya.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 or VOIP in G350 media gateway. In this configuration, the shuffling is turn off.
- 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 SingTel only and does not apply to any IP phone (H.323 or SIP) within the enterprise.
- In page 3, the Source to Destination Region 6 **Codec Set** is set as default to the codec set in page 1 of the form. The Source Region 6 to Destination Region 1 **Codec Set** is set as 6 for us to specify the codec to be used for SIP Trunk to the enterprise site.

In this case, the SIP trunk is assigned to different IP network region as the G350 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 SingTel's SIP Trunking service. These parameters may need to be aligned with the specific values provided by SingTel.

```
display ip-network-region 6                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 6
Location:          Authoritative Domain: singtel-sip.com
    Name: Alctel region
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: no
      Codec Set: 6              Inter-region IP-IP Direct Audio: no
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 65535
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 6                                     Page   3 of  19

                   Inter Network Region Connection Management

 src dst codec direct   WAN-BW-limits   Video                          Dyn
 rgn rgn  set   WAN  Units    Total Norm  Prio Shr Intervening-regions  CAC IGAR
 6   1    6     y    NoLimit                                                 n
 6   2
 6   3
 6   4
 6   5
 6   6    6
 6   7
 6   8
 6   9
 6   10
 6   11
 6   12
 6   13
 6   14
 6   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 SingTel, the codec G.729A is supported for On-Net calls and G.729B, G.711A and G.711Mu is supported for Off-Net calls. 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 6                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 6

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.729B           n          2        20
 2: G.729A           n          2        20
 3: G.711MU          n          2        20
 4: G.711A           n          2        20
 5:
 6:
 7:


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

```
display ip-codec-set 6                                          Page   2 of   2

                          IP Codec Set

                              Allow Direct-IP Multimedia? n



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

Figure 8: IP Codec Set Form

Step 5: Configure the Signaling Groups

For interoperability with SingTel, two signaling groups must be configured. One signaling group will be used for inbound and outbound calls while the second signaling group will be used for inbound anonymous call. The second signaling group is to provide for anonymous caller with no proper caller's domain/IP address in the SIP Invite messages. 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, sometimes 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. In order for inbound calls to then be routed, 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 first **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 "Singtel-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 SingTel proxy in the **Far-end Domain** field. In this configuration, the domain name is not used. Rather, the ip address of the Singtel ConnectPLUS Gatekeeper is used as no DNS server is provided. This ip address 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.

- **Direct IP-IP Audio Connections** field must be set to 'n' as shuffling is not supported
- 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 2
                                SIGNALING GROUP

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








   Near-end Node Name: procr                  Far-end Node Name: SingTel-SES
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 6
       Far-end Domain: 10.6.0.6

                                              Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? n
                                                         IP Audio Hairpinning? n
 Enable Layer 3 Test? n
 Session Establishment Timer(min): 120
```

Figure 9: Outbound Signaling Group Form

Next, configure the second **Signaling Group** form following the same steps used for the first signaling group above with one exception; leave the **Far-end Domain** field blank as shown in **Figure 10**.

```
add signaling-group 3
                                SIGNALING GROUP

 Group Number: 3              Group Type: sip
                        Transport Method: tls





   Near-end Node Name: procr                  Far-end Node Name: SingTel-SES
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 6
       Far-end Domain:

                                              Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? n
                                                         IP Audio Hairpinning? n
 Enable Layer 3 Test? n
 Session Establishment Timer(min): 3
```

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 and inbound signaling group 2 and the other with the inbound signaling group 3.

Configure the *first* **Trunk Group** form as shown in **Figure 11** 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 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 SingTel. Calls involving a SIP endpoint and SingTel 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 to SingTel        COR: 95       TN: 1        TAC: 401
   Direction: outgoing       Outgoing Display? n
 Dial Access? n
Queue Length: 0
Service Type: tie

                                                       Signaling Group: 2
                                                     Number of Members: 10

display trunk-group 2                                                  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): 900
add trunk-group 2                                                  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): 900
```

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

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

- Set the **Preferred Minimum Session Refresh Interval(sec)** field to 900 seconds. This field specifies the refresh INVITE Timer sent to the far-end. Avaya Communication Manager sends a session refresh request as a Re-INVITE or UPDATE after every timer interval to maintain ongoing sessions. As there are gateways on the network that are affected by this timer, the recommended minimum value for this is "900" to avoid unnecessary SIP Messaging for session negotiation.

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.

On Page 4 of the **Trunk Group** form:

- Set the **Telephone Event Payload Type** field to 127 as asymmetric telephony event negotiation is supported for On-Net calls.

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



                     Numbering Format: public
                                                UUI Treatment: service-provider


                                                Replace Unavailable Numbers? N
```

```
add trunk-group 2                                             Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n

             Telephone Event Payload Type: 127
```

Figure 12: Trunk Group Form (Both way) – Page 3 & 4

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 3. 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 SingTel. Calls involving a SIP endpoint and SingTel will use two trunk members for the duration of the call.

```
add trunk-group 3                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 3                   Group Type: sip          CDR Reports: y
  Group Name: SIP Trunk - Anonymous      COR: 95       TN: 1        TAC: 403
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                     Signaling Group: 3
                                                   Number of Members: 10
display trunk-group 3                                            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): 900
```

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.

On Page 4 of the **Trunk Group** form:

- Set the **Telephone Event Payload Type** field to 127. as asymmetric telephony event negotiation is supported for On-Net calls.

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



                     Numbering Format: public
                                                 UUI Treatment: service-provider


                                                 Replace Unavailable Numbers? n
add trunk-group 3                                            Page   4 of  21
                              PROTOCOL VARIATIONS

                       Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n

             Telephone Event Payload Type: 127
```

Figure 14: Trunk Group Form (Inbound) – Page 3 & 4

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 2 should send the calling party number 2XXX when an outbound call uses SIP trunk Group #2 (*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.

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
Ext Ext              Trk       CPN          CPN
Len Code             Grp(s)    Prefix       Len
                                                     Total Administered: 1
 5  2                2                      5           Maximum Entries: 240
```

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 SingTel SIP Trunking Service to any destination.

Use the **change dialplan analysis** command to add **0** 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
        0          1   fac
        2          5   ext
        4          3   dac
        85         7   ext
        *          3   fac
        #          3   fac
```

Figure 16: Change Dialplan Analysis Form

Use the **change feature-access-codes** command to specify **0** 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:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code:
    Auto Route Selection (ARS) - Access Code 1: 0     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA:        All:        Deactivation:
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
                   Contact Closure   Open Code:          Close 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 “0”. In this sample configuration, the PSTN numbers dialed begins with 01 and the call is to be routed to a route pattern containing the SIP trunk groups used for SingTel as in **Figure 18**. 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
    01                   7    24    3         intl           n
                                                             n
                                                             n
                                                             n
                                                             n
                                                             n
                                                             n
                                                             n
                                                             n
                                                             n
                                                             n
                                                             n
                                                             n
                                                             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 3 will be used to route calls to trunk group 2, (the SIP trunk created in Step 6**, Figure 11**).

```
change route-pattern 3                                          Page   1 of   3
                    Pattern Number: 3   Pattern Name: To Sintel - IDD
                             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: 2    0                        0                                     n   user
 2:                                                                     n   user
 3:                                                                     n   user
 4:                                                                     n   user
 5:                                                                     n   user
 6:                                                                     n   user

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

Figure 19: Route Pattern Form

Step 9: 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 SingTel 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 20**.
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 or One-X 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 20003                                               Page   1 of   5
                                     STATION

Extension: 20003                         Lock Messages? n               BCC: 0
     Type: 4608D+                        Security Code:                  TN: 1
     Port: X                           Coverage Path 1:                 COR: 1
     Name: SIP 9630                    Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 20003
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english                 Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n

                                            Customizable Labels? y
```

Figure 20: 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 20003                                               Page   2 of   5
                                     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
                                                         EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number? y
       Service Link Mode: as-needed
         Multimedia Mode: enhanced                 Audible Message Waiting? n
    MWI Served User Type:                       Display Client Redirection? n
              AUDIX Name:                      Select Last Used Appearance? n
                                                 Coverage After Forwarding? s
                                                   Multimedia Early Answer? n
                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 20003        Always Use? n IP Audio Hairpinning? n
```

Figure 21: Station Administration – Page 2

On Page 4 of the **Station** form, configure 5 call appearances under the **Button Assignments** section for the SIP telephone, as shown in **Figure 22**.

```
add station 20003                                               Page   4 of   5
                                     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 22: Station Administration – Page 4

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 23**:

- 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 ‘1’, which is the number assigned to the 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 20003                  Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk      Config
 Extension                  Prefix                       Selection  Set
 20003            OPS            -     20003              1          1
                                 -
```

Figure 23: 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 20003                   Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls        Bridged
 Extension     Limit      Mode         Allowed      Calls
 20003          5       both           all          both
```

Figure 24: 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 SingTel's 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 SingTel

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 25**.

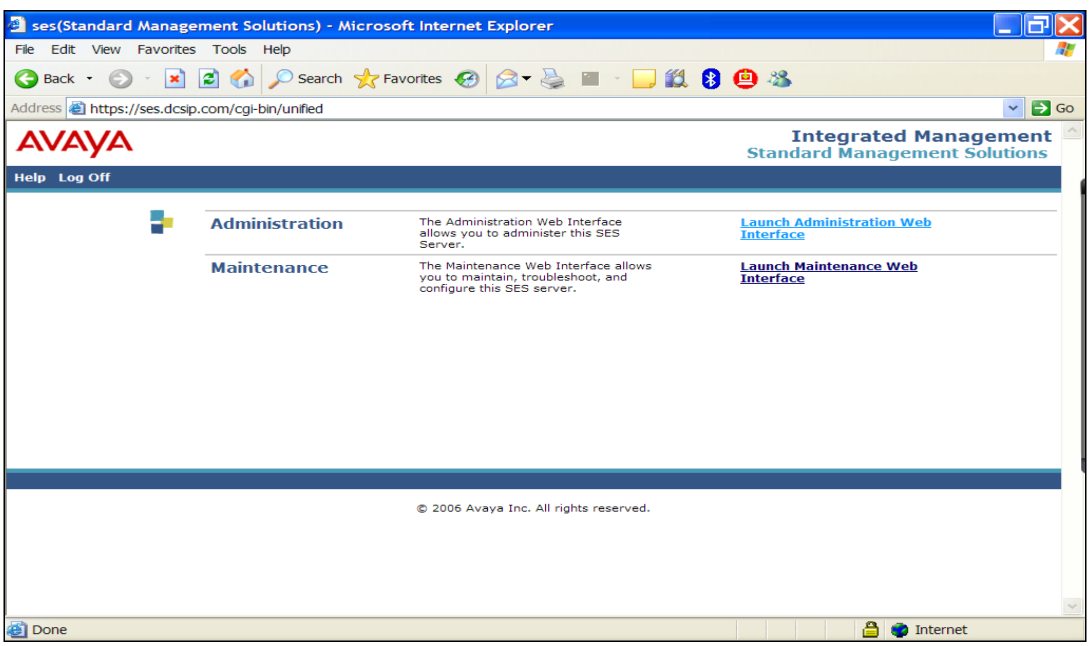

Figure 25 - Avaya SES Main Screen

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 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 "singtel-sip.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** or the Avaya SES server address which in this configuration is "10.6.0.26" unless the WebLM server is not co-resident with this server.
- After configuring the **System Properties** screen, shown in **Figure 26**, click the **Update** button.

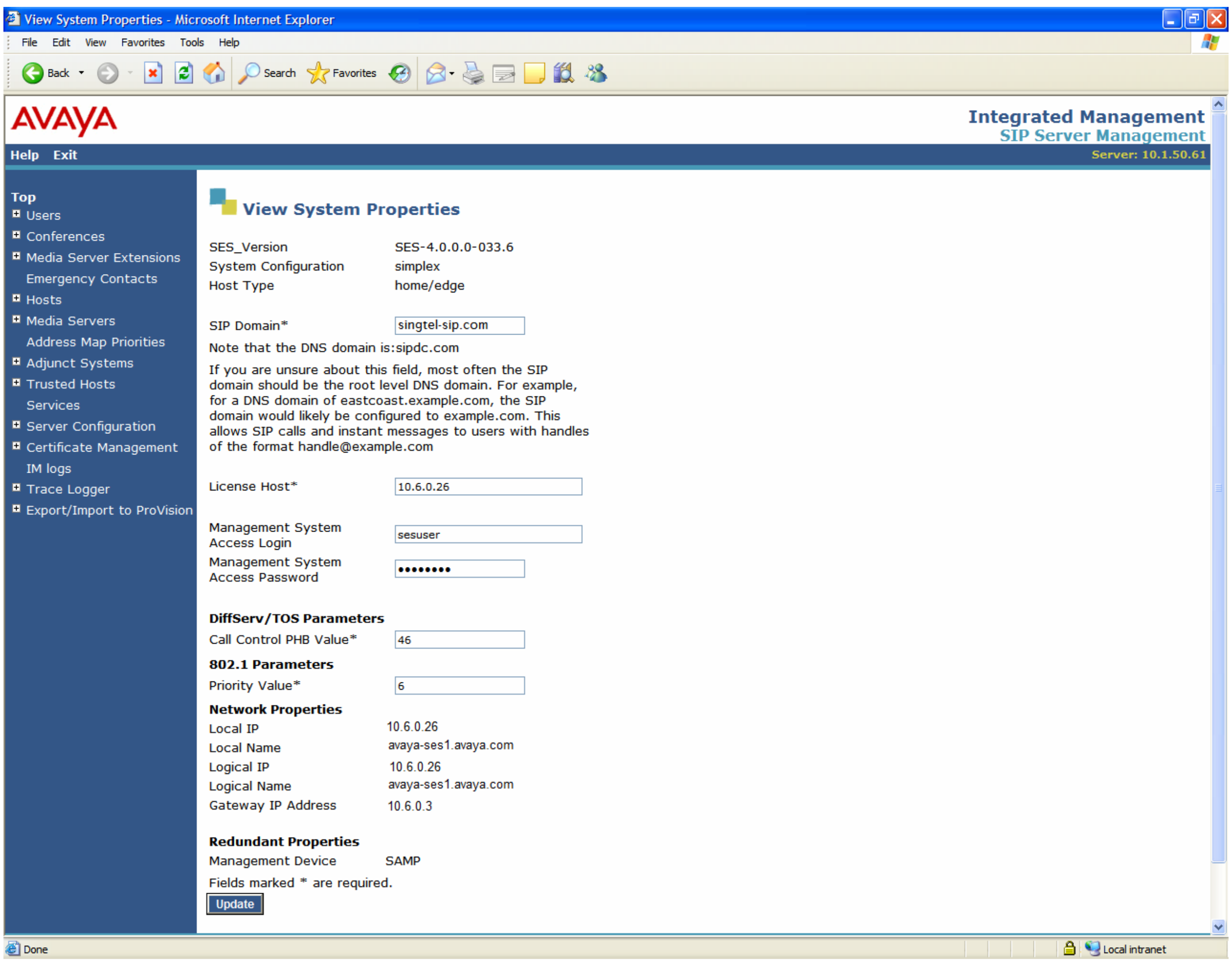

Figure 26: System Properties

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 27** 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 27**) 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 27**.
- Click the **Update** button.

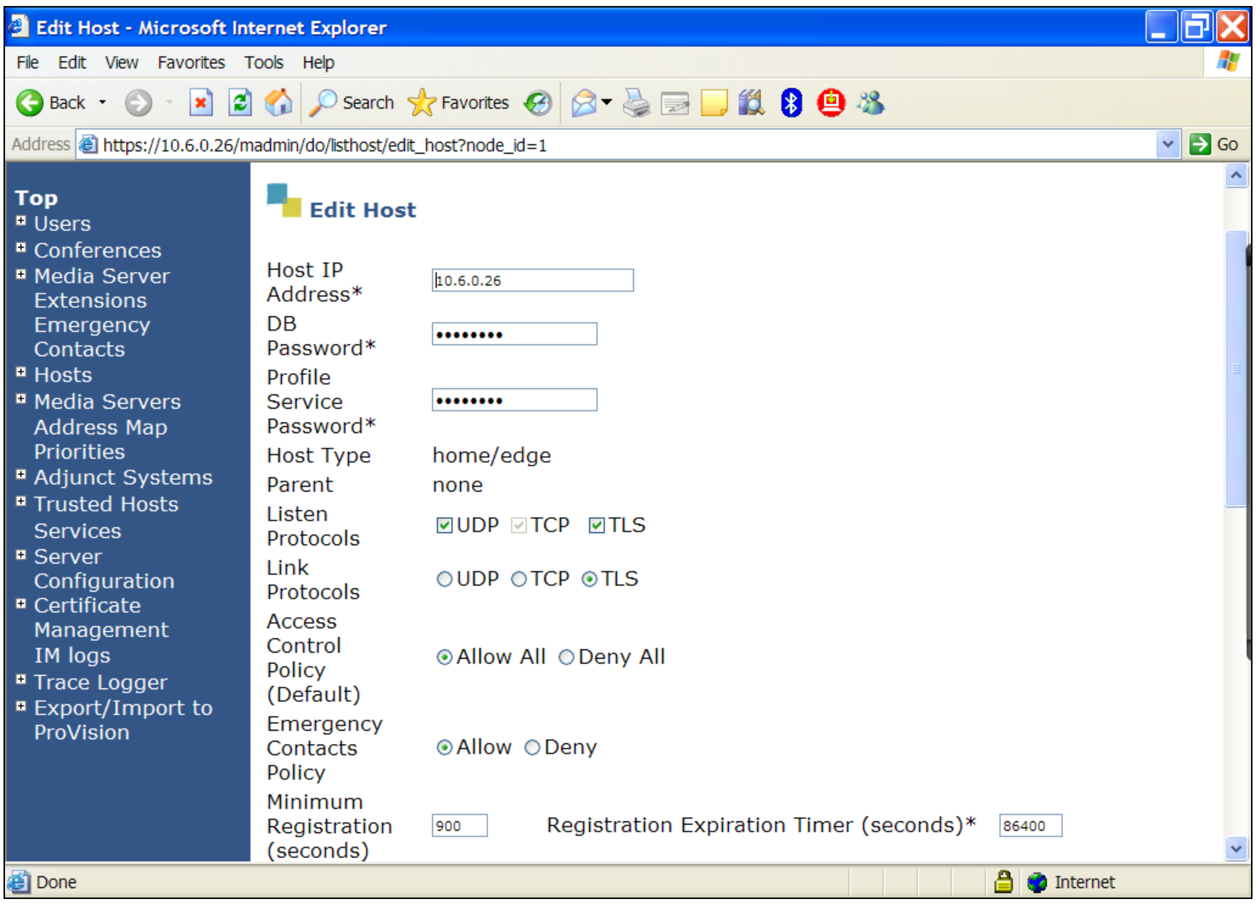

Figure 27: Edit Host

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., Singtel-SES).
- 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.)
- Enter the same address for **Media Server Admin Address**.
- Enter the **Media Server Admin Login** provided for SES access to Avaya Communication Manager and the **Media Server Admin Password**.
- After completing the **Add Media Server** screen, click on the **Add** button.

Figure 28 below shows the edited Media Server Interface.

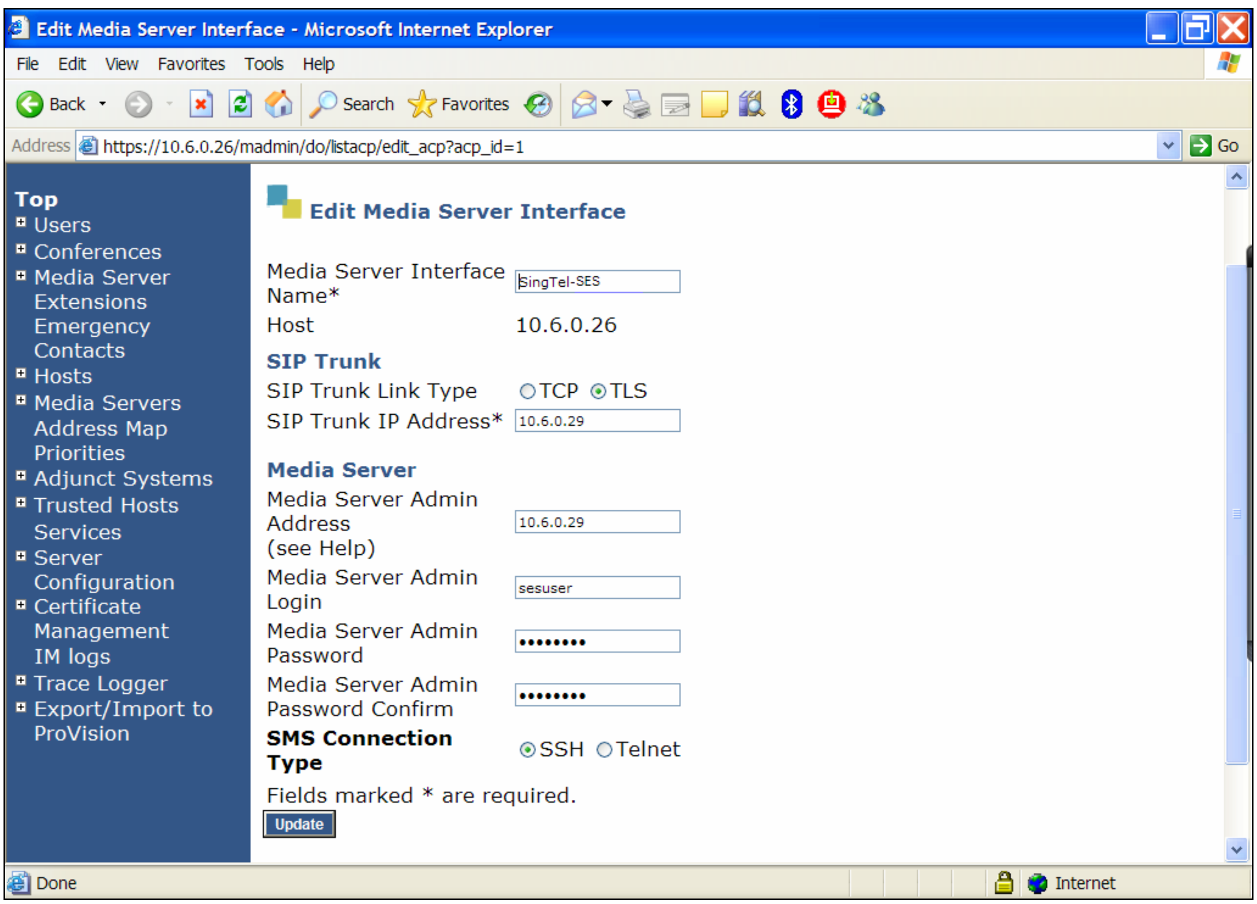

Figure 28: Media Server Interface

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 On-Net 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 Server extension to that phone. Address map definitions for SIP endpoints not assigned a server extension and connections to multiple service providers are beyond the scope of these Application Notes.

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

```
INVITE sip:20005@10.6.0.6:5060 SIP/2.0
```

The user portion in this case is the 5 digit number "20005". 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 SingTel. 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 **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 numbers provided by SingTel are 2XXXX. The pattern specification (without the double quotes) for numbers assigned is: "^sip:2000[1-9]{1}".
- Click the **Add** button once the form is completed.
- The screen shown in **Figure 29** shows the entries.

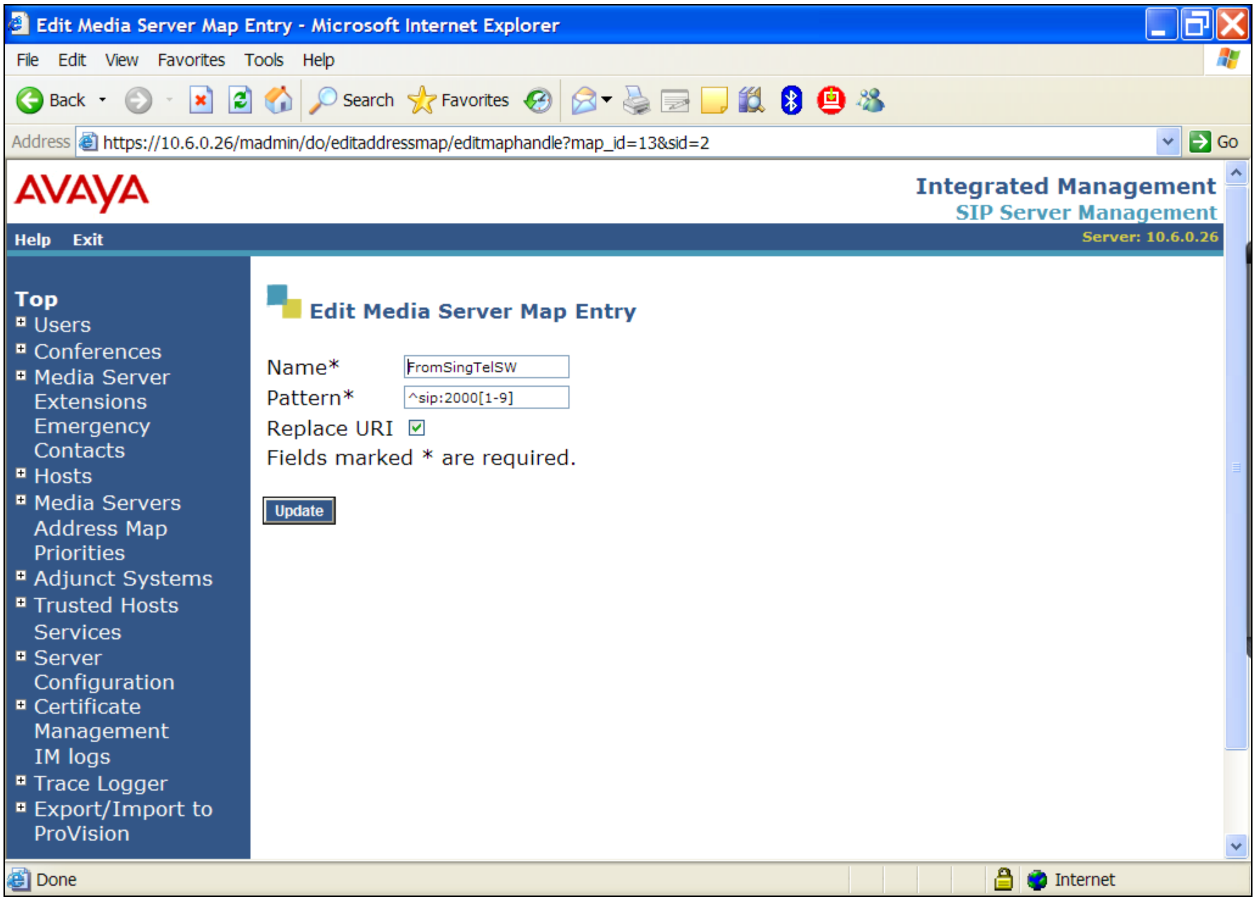

Figure 29: Media Server Address Map

After configuring the media server address map, the **List Media Server Address Map** screen appears as shown in **Figure 30**.

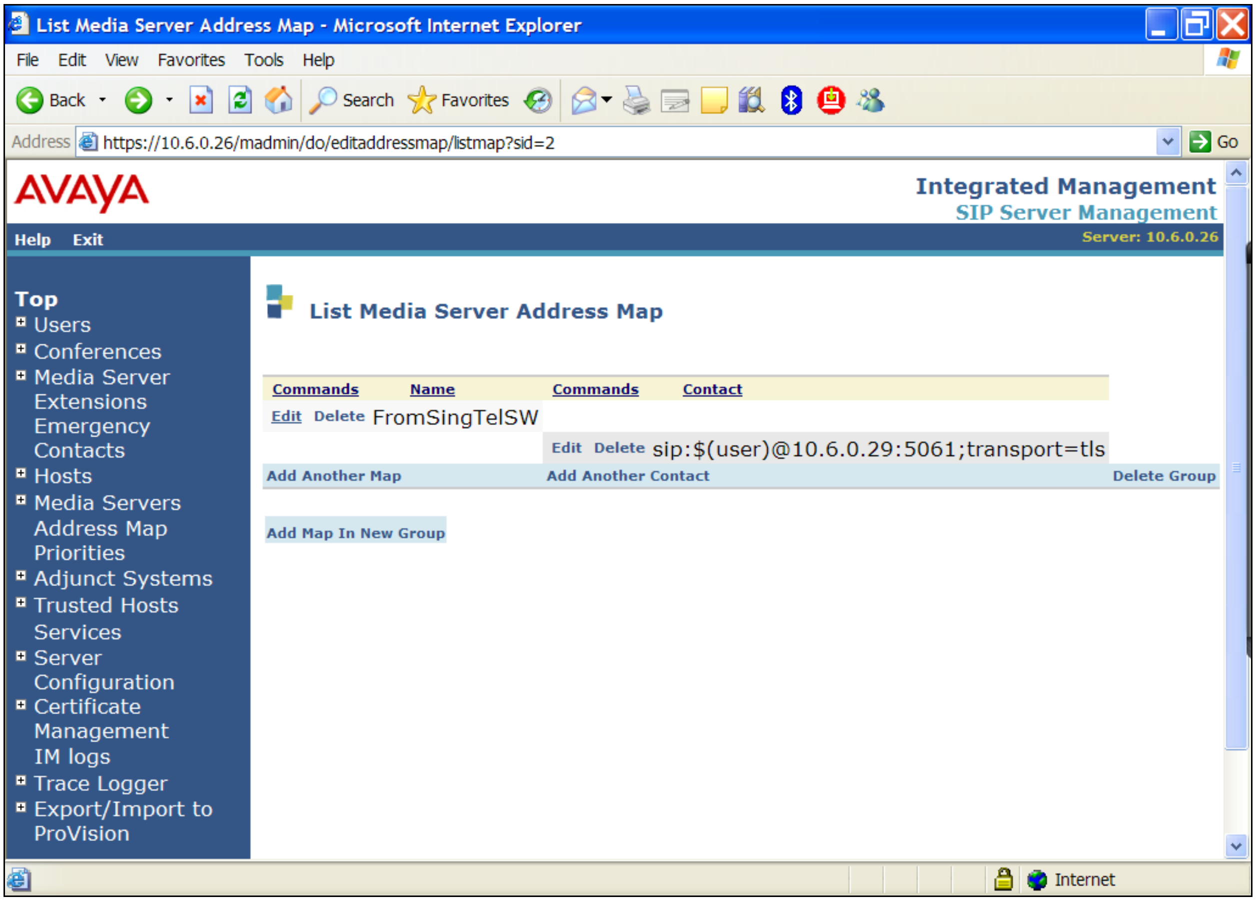

Figure 30: List Media Server Address Map

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 30**, the following contact was created:

sip:$(user)@10.6.0.29:5061;transport=tls

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

Step 6: Specify Address Maps to SingTel

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 SingTel'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 SingTel gatekeeper 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 SingTel SIP signaling gatekeeper). In this configuration, the Avaya SES routing rule for the SIP trunk group will be to send all outbound traffic to SingTel's SIP Trunking Service.

To perform this, several dialing pattern will be created in the Avaya SES.
• An example is the pattern (without the double quotes) of "`^sip:85[0-9]{5}`" will match on all sip calls having digits beginning with 85.

Note that additional or more specific pattern matches would be used if necessary to selectively route SIP traffic to different destinations. Also note that a user dialed access code (such as 0 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 is shown in **Figure 32**.

- 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**. Enter a descriptive name for the map, such as "OUTSingtel-Ext85"
- Specify an appropriate pattern for the call type. In this example, the pattern used is "^sip:85[0-9]{5}".
- Leave the **Replace URI** checkbox selected.
- Click the **Add** button.

The screen in **Figure 31** shows the entries on the Host Map.

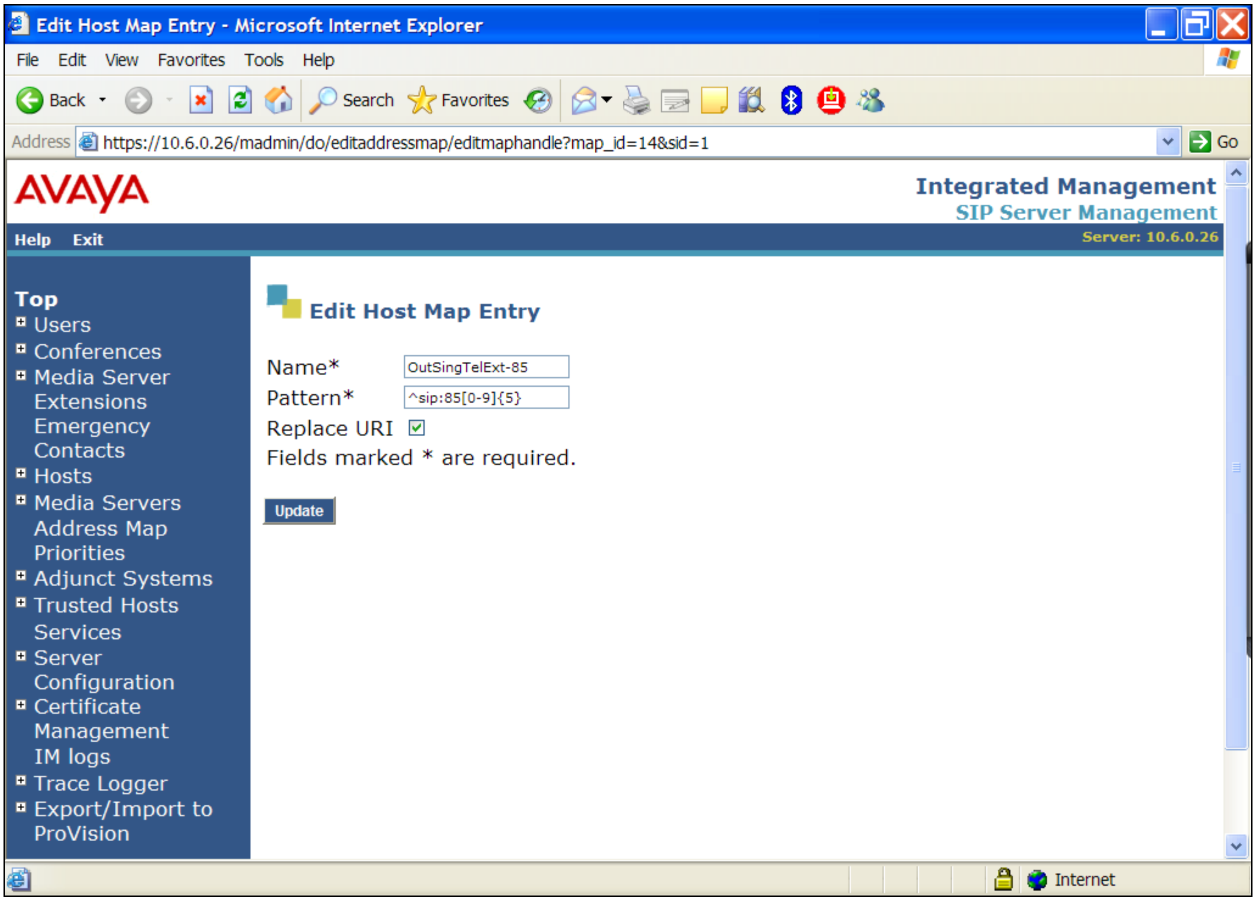

Figure 31: Edit Host Map Entry

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

Step 7: Specify the SingTel SIP Gatekeeper Information

The next step is to enter the contact address for the SingTel SIP Gatekeeper. In this example, an IP address is used to identify SingTel's SIP Gatekeeper. The customer's specific information will be provided by SingTel.

To enter the SingTel SIP Gatekeeper 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 31** 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)@10.6.0.6:5060;transport=udp

  The user part in the original request URI is inserted in place of the "$(user)" string before the message is sent to SingTel.
- 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 32**.

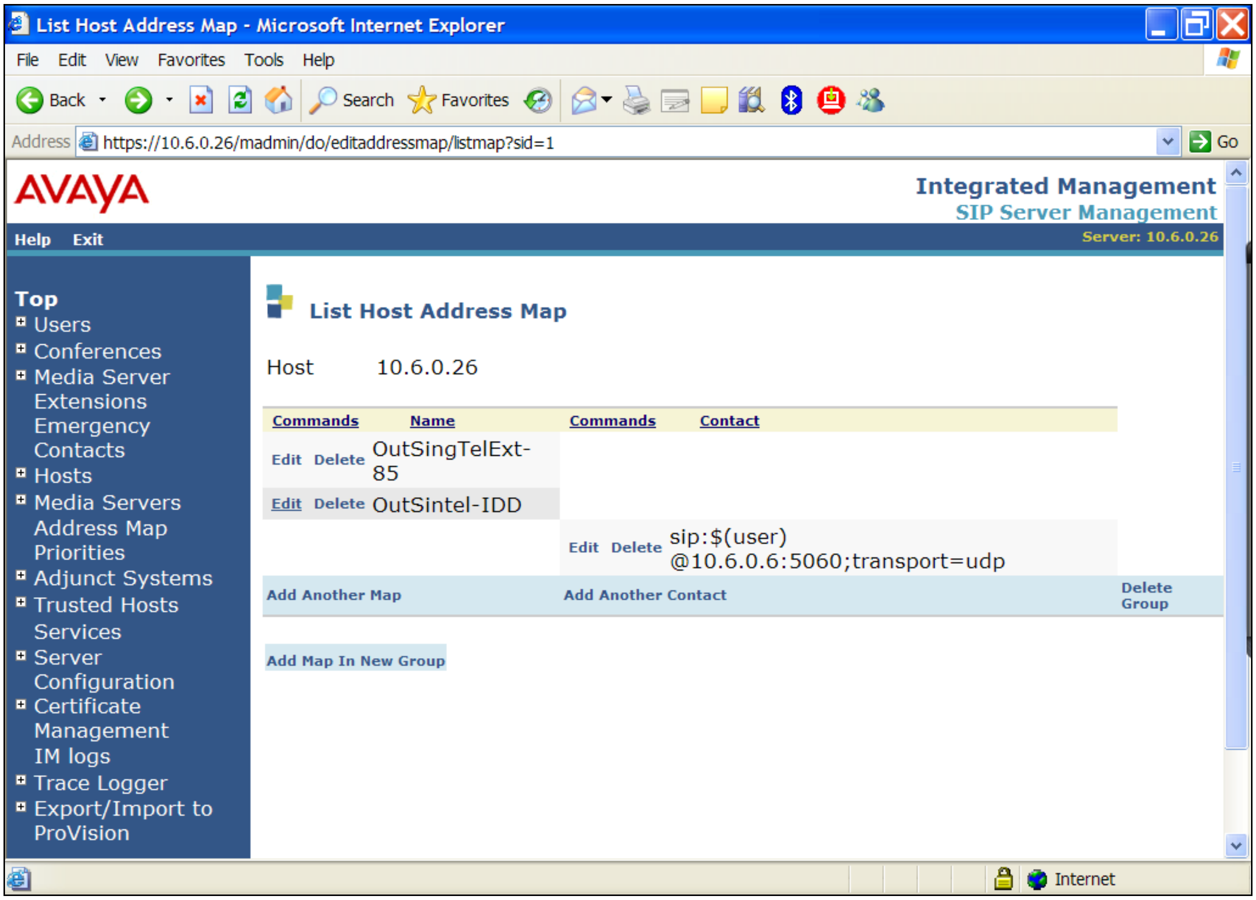

Figure 32: List Host Address Map

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 33**.

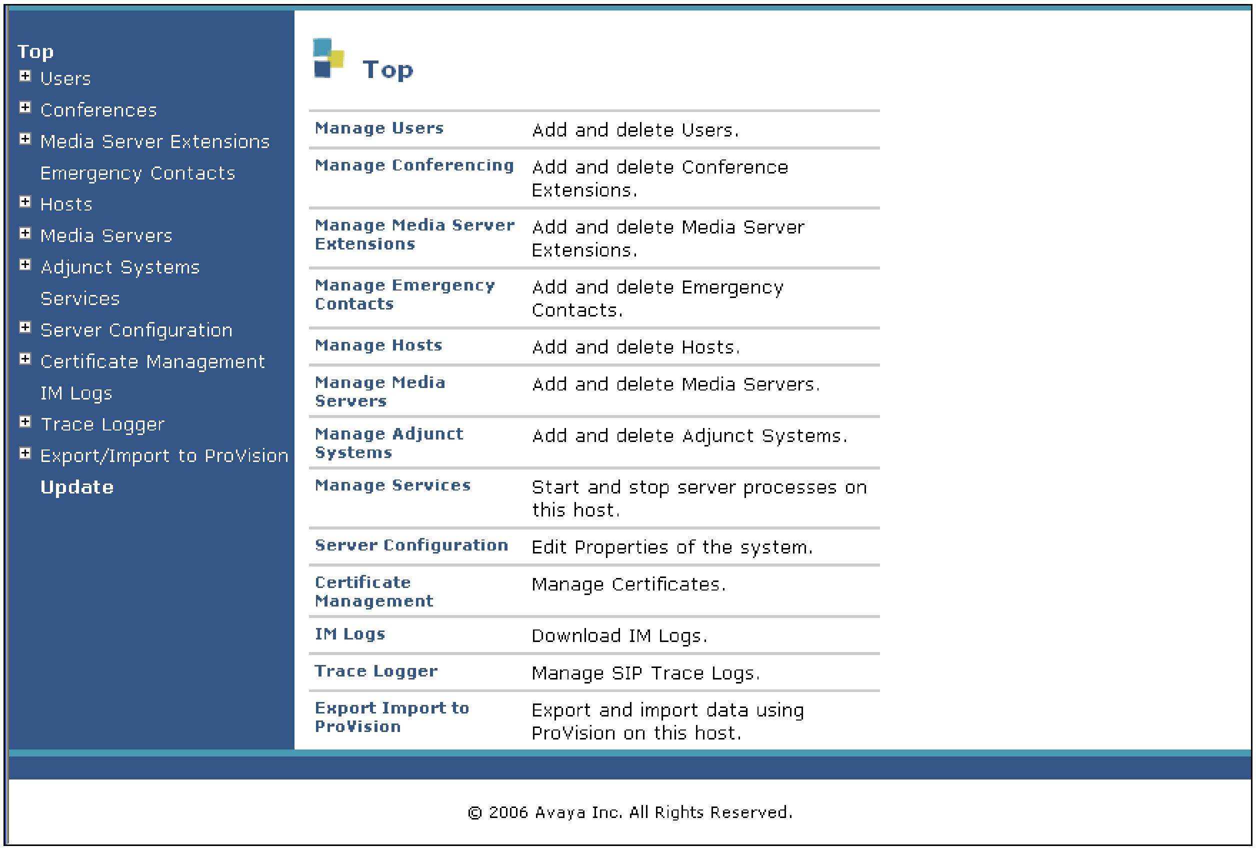

Figure 33: Update Following Avaya SES Administrative Changes

Step 9: Specify the SingTel SIP Gatekeeper as a Trusted Host

The final step to complete the SIP trunk administration on Avaya SES is to designate the IP address of SingTel SIP Gatekeeper as a trusted host. As a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the designated IP address. If multiple SIP proxies are used, the IP address of each SIP proxy must be added as a trusted host.

To configure a Trusted Hosts:

- Select **Trusted Hosts** in the left pane of the Administration web interface. This will display the **Manage Trusted Hosts** screen.
- Click on the **Add Managed Trusted Hosts** link. Enter the Singtel SIP gatekeeper IP Address as the Trusted Host which in this configuration is “10.6.0.6”.
- Select the **Host** which is the Avaya SES server “10.6.0.26”.
- Press the **Add** button. This will cause a confirmation screen to appear.
- Press **Continue** on the confirmation screen.

Figure 34 illustrates the results of the Trusted Hosts entry.

- Complete the trusted host configuration by clicking on the **Update** link.

  If the **Update** link is not visible, refresh the page by selecting **Top** from the left hand menu.

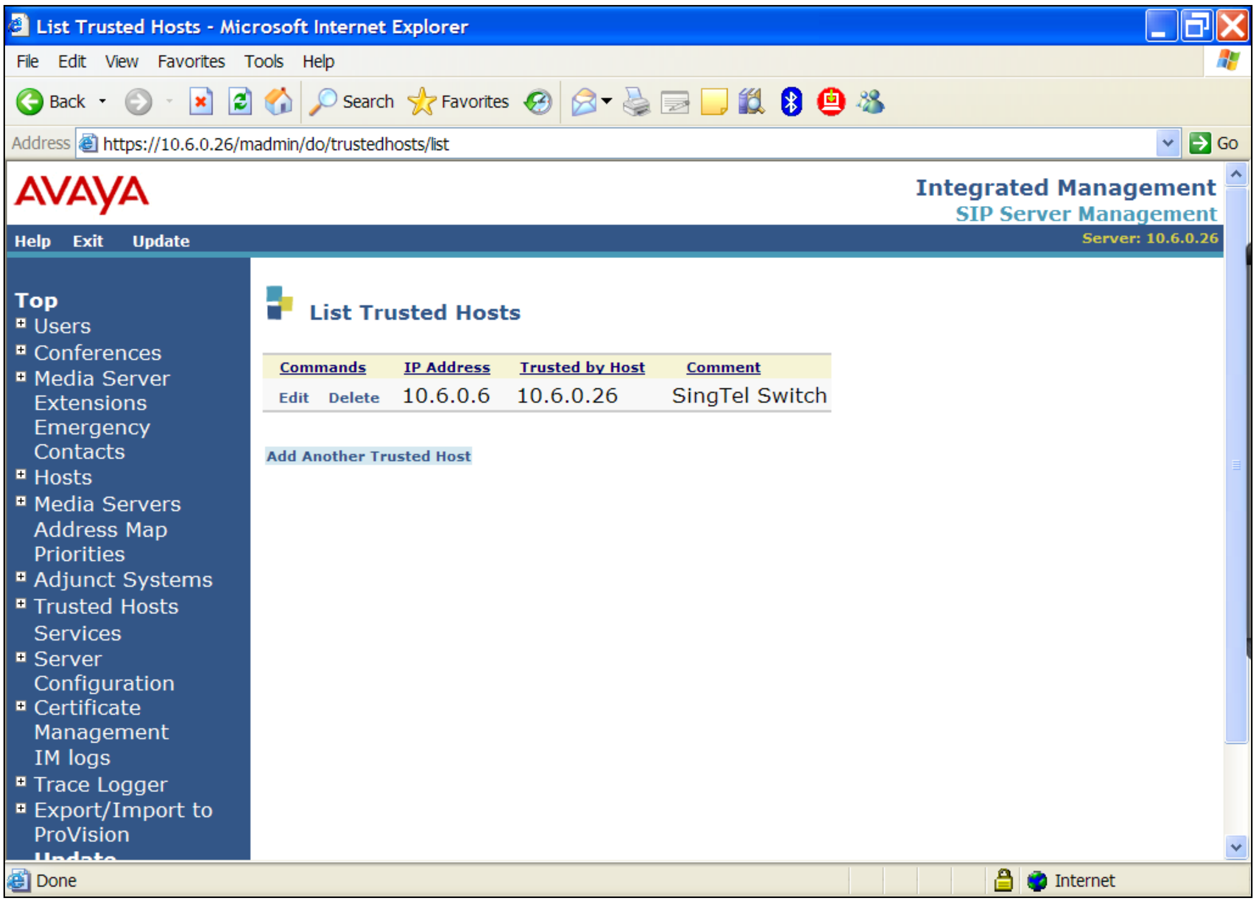

Figure 34: Configuring a Trusted Host

4.2. Configuration for SIP Telephones

This section provides basic instructions for completing the administration necessary to support the optional Avaya 46xx SIP or 96xx 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 (10.6.0.26) 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.

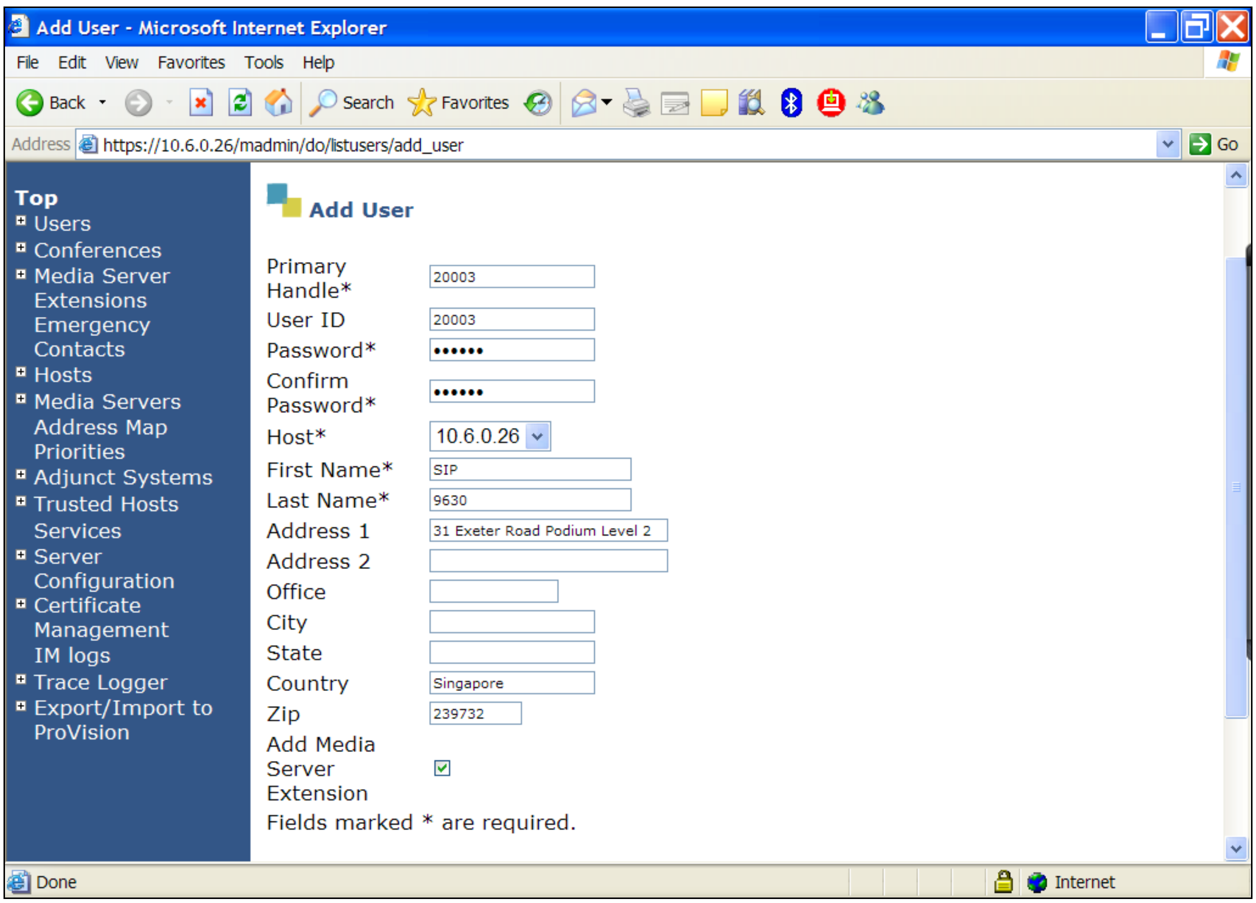

Figure 35: Add User

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 20**, 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.

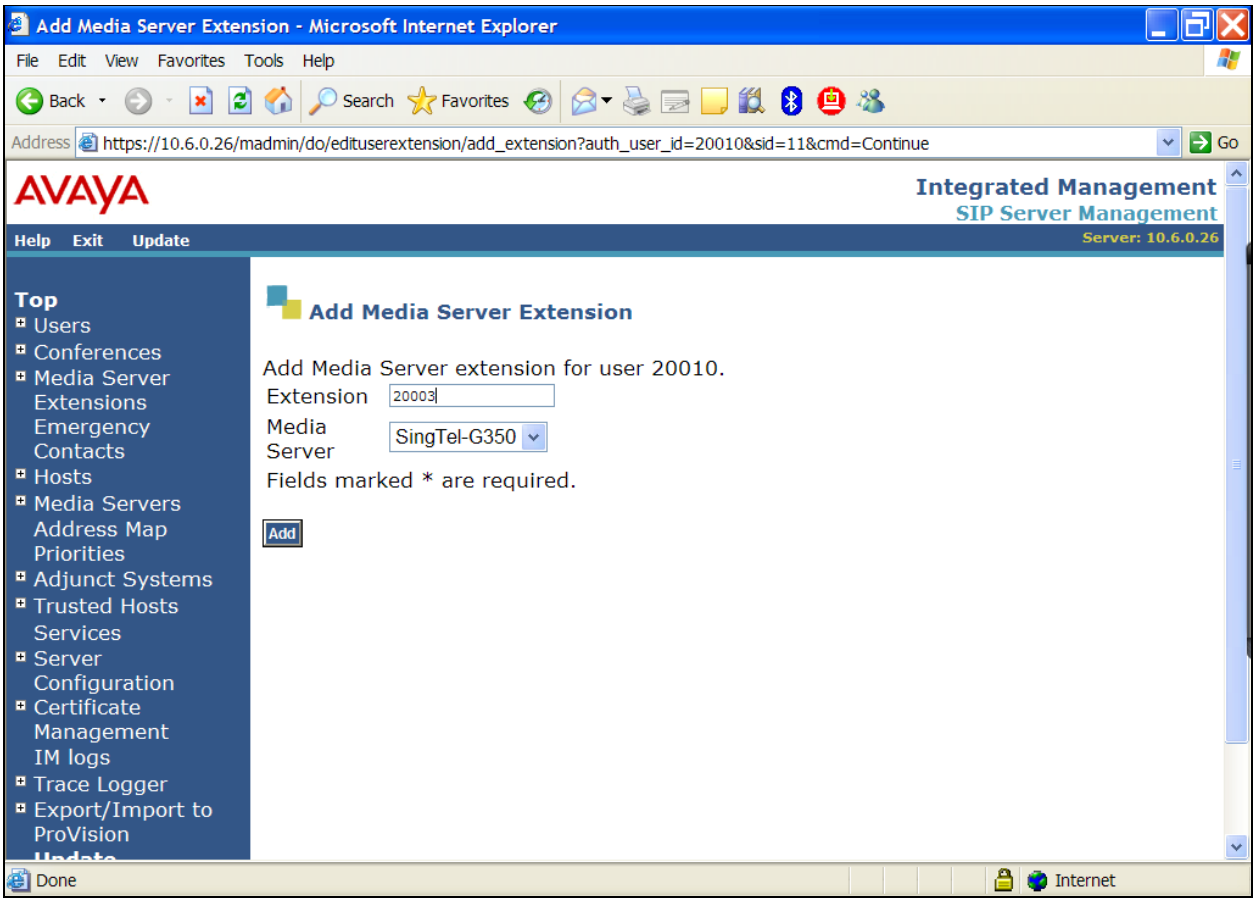

Figure 36: Add Media Server Extension

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

5. SingTel SIP Trunking Services Configuration

In order to use SingTel VoIP Services, a customer must order service from SingTel using the SingTel sales processes. SingTel local contact is at ***1800 7383838*** or +***65 7383838.*** The email contact address is ***businessco@singtel.com.***

6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP Trunking interoperability between SingTel's ConnectPlus 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

The enterprise site was configured to use the commercially available SIP Trunking Service provided by SingTel. This allowed the enterprise site to use SIP Trunking for On-Net or PSTN calling. As the Alcatel softswitch is a Class 4 switch, not many enterprise features can be interworked.

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

- Incoming calls to the enterprise site from On-Net calls were routed to the numbers assigned by SingTel.
- Outgoing calls from the enterprise site were completed via SingTel to the On-Net users or PSTN destinations.
- Calls using SIP, H.323, digital or analog endpoints supported by the Avaya IP telephony solution.
- Calls using G.711 codec and G.729.
- T.38 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 call hold.

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
T.38 Fax Outgoing to PSTN	Outgoing FAX to PSTN was rejected by PSTN gateway.	Patch fix will be released in Service Pack 3 for CM4.0.1.
DTMF over IP using RFC 2833	DTMF sending via RTP payload with Telephone event is not accepted	The Singtel ConnectPLUS Networks needs to support it as there is no workaround solution.

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 or analog endpoints can place outbound and receive inbound PSTN calls through SingTel.

1. Verify that endpoints at the enterprise site can place calls to the PSTN or On-Net 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 On-Net users and that the call can remain active for more than 35 seconds.

3. Verify that the user on the PSTN or On-Net 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 SingTel's SIP Trunking Service, contact SingTel Account Manager assigned by SingTel.

9. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya Communication Manager and Avaya SES telephony solution to SingTel's SIP Trunking Service on the ConnectPlus IP VPN. SIP Trunking uses the Session Initiation Protocol (SIP) to connect private company networks to the ConnectPlus VPN 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*, February 2007, Issue 3, Document Number 03-300509.
[2] *Feature Description and Implementation for Avaya Communication Manager,* February 2007, Issue 5, Document Number 555-245-205
[3] *Avaya Extension to Cellular User Guide Release 4.0,* February 2007, Issue 10, Document Number 210-100-700.
[4] *Installing and Administering SIP Enablement Services,* May 2007, Issue 4.0, Document Number 03-600768
[5] *SIP Support in Avaya Communication Manager Running on the Avaya S8300,S8400, S8500 series and S8700 series Media Server*, May 2007, Issue 7, 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 SingTel 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 SingTel to Avaya SIP Enablement Services:

```
Session Initiation Protocol
    Request-Line: INVITE sip:20005@10.6.0.6:5060 SIP/2.0
    Message Header
        Via: SIP/2.0/UDP  10.6.0.18:5060;branch=z9hG4bK1A249D
        From: <sip:6500800@10.6.0.18>;tag=E149E2C-274
        To: <sip:20005@10.6.0.6>
        Date: Sun, 03 Mar 2002 17:37:12 GMT
        Call-ID: 20FC3DDF-2E0411D6-803EA395-AFB4FF7D@10.6.0.18
        Supported: 100rel,timer,replaces
        Min-SE:  1800
        Cisco-Guid: 466361722-772018646-2151392149-2947874685
        User-Agent: Cisco-SIPGateway/IOS-12.x
        Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE, NOTIFY, INFO,
UPDATE, REGISTER
        CSeq: 101 INVITE
        Max-Forwards: 70
        Remote-Party-ID: <sip:6500800@10.6.0.18>;party=calling;screen=no;privacy=off
        Timestamp: 1015177032
        Contact: <sip:6500800@10.6.0.18:5060>
        Expires: 60
        Allow-Events: telephone-event
        Content-Type: application/sdp
        Content-Length: 229
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): CiscoSystemsSIP-GW-UserAgent 1793 5320 IN IP4
10.6.0.18
            Session Name (s): SIP Call
            Connection Information (c): IN IP4 10.6.0.18
            Time Description, active time (t): 0 0
            Media Description, name and address (m): audio 16838 RTP/AVP 18 19
            Connection Information (c): IN IP4 10.6.0.18
            Media Attribute (a): rtpmap:18 G729/8000
            Media Attribute (a): fmtp:18 annexb=no
            Media Attribute (a): rtpmap:19 CN/8000
            Media Attribute (a): ptime:40
```

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

```
Session Initiation Protocol
    Request-Line: INVITE sip:0016591773604@10.6.0.6 SIP/2.0
    Message Header
        Call-ID: 80607f853d89dc1c13472ca5600
        CSeq: 1 INVITE
        From: "IP H.323" <sip:20005@singtel-sip.com:5061>;tag=80607f853d89dc1c03472ca5600
        Record-Route: <sip:10.6.0.26:5060;lr>,<sip:10.6.0.29:5061;lr;transport=tls>
        To: "0016591773604" <sip:0016591773604@10.6.0.6>
        Via: SIP/2.0/UDP 10.6.0.26:5060;branch=z9hG4bK838383030303636344ce.0,SIP/2.0/TLS
10.6.0.29;psrrposn=2;received=10.6.0.29;branch=z9hG4bK80607f853d89dc1c23472ca5600
        Content-Length: 172
        Content-Type: application/sdp
        Contact: "IP H.323" <sip:20005@10.6.0.29:5061;transport=tls>
        Max-Forwards: 63
        User-Agent: Avaya CM/R014x.00.1.731.2
        Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
        History-Info: <sip:0016591773604@10.6.0.6>;index=1
        History-Info: "0016591773604" <sip:0016591773604@10.6.0.6>;index=1.1
        Supported: 100rel,timer,replaces,join,histinfo
        Min-SE: 1800
        Session-Expires: 1800;refresher=uac
        P-Asserted-Identity: "IP H.323" <sip:20005@singtel-sip.com:5061>
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 1 1 IN IP4 10.6.0.29
            Session Name (s): -
            Connection Information (c): IN IP4 10.6.0.28
            Time Description, active time (t): 0 0
            Media Description, name and address (m): audio 2056 RTP/AVP 18 127
            Media Attribute (a): rtpmap:18 G729/8000
            Media Attribute (a): fmtp:18 annexb=yes
            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: 0016591773604@10.6.0.6;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 DevConnect Program at devconnect@avaya.com.