Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between SingTel SIP Trunking Service on the Meg@POP IP VPN 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 SingTel Meg@POP IP VPN Network and an Avaya IP Telephony solution. The solution consists of Avaya Communications Manager, Avaya SIP Enablement Services, and various Avaya SIP, H.323 and analog endpoints. The Avaya Communication Manager connects to the SingTel Meg@POP Network via direct SIP Trunking. The Avaya SIP Enablement Services is included to support the Avaya SIP endpoints only.

SingTel is one of the main service providers in Singapore. Certification of SingTel's SIP Trunking Service will allow existing or new customers on this domestic network to have SIP trunks to their Avaya PBXs other than the traditional type of trunks.

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

1. Introduction

These Application Notes describe the steps for configuring SIP Trunking between the SingTel SIP Trunking Service on the SingTel Meg@POP IP VPN Network and an Avaya IP telephony solution. The Avaya solution consists of Avaya Communication Manager, Avaya SIP Enablement Services (SES), and Avaya SIP, H.323 and analog endpoints. The Avaya Communication Manager connects to the SingTel Meg@POP Network via direct SIP Trunking. The Avaya SIP Enablement Services is included to support the Avaya SIP endpoints only.

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1.1. Interoperability Compliance Testing

A simulated enterprise site consisting of an Avaya IP telephony solution supporting SIP Trunking was connected to the SingTel Meg@POP Network using a dedicated IP connection. The enterprise site was configured to use the commercially available SIP Trunking Service provided by SingTel. This allowed the enterprise site to leverage SingTel IP Centrex Services through SIP Trunking Service and call termination to PSTN.

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 SingTel.
- Outgoing calls from the enterprise site were completed via SingTel to the PSTN destinations.
- Calls placed and received using Avaya SIP, H.323 or analog telephones.
- Calls placed and received using G.729A and G.711 codecs.
- G.711 codec used for outgoing fax. G.729A and G.711 codecs used for incoming fax.
- DTMF tone transmission using RFC 2833 with successful voice mail navigation.
- Telephone features such as call hold, transfer, call forwarding, conference, call pickup and calling number display.

1.2. Support

For technical support on SingTel SIP Trunking Service on the SingTel Meg@POP IP VPN Network, contact the SingTel Account Manager assigned by SingTel or dial 1800-763-1111 for general enquiries.

2. Reference 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.

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

- An Avaya S8300B Server with an Avaya G700 Media Gateway. The S83000B served as the host processor for Avaya Communication Manager.
- Avaya SES software operating on an Avaya S8500B server.
- Avaya 9600 Series IP Telephones (configured to use either the SIP or H.323 protocol) and Avaya 6221 Analog telephones.

In this configuration, Avaya Communication Manager and Avaya SES are setup with private IP addresses behind a router on the SingTel Meg@POP IP VPN Network.

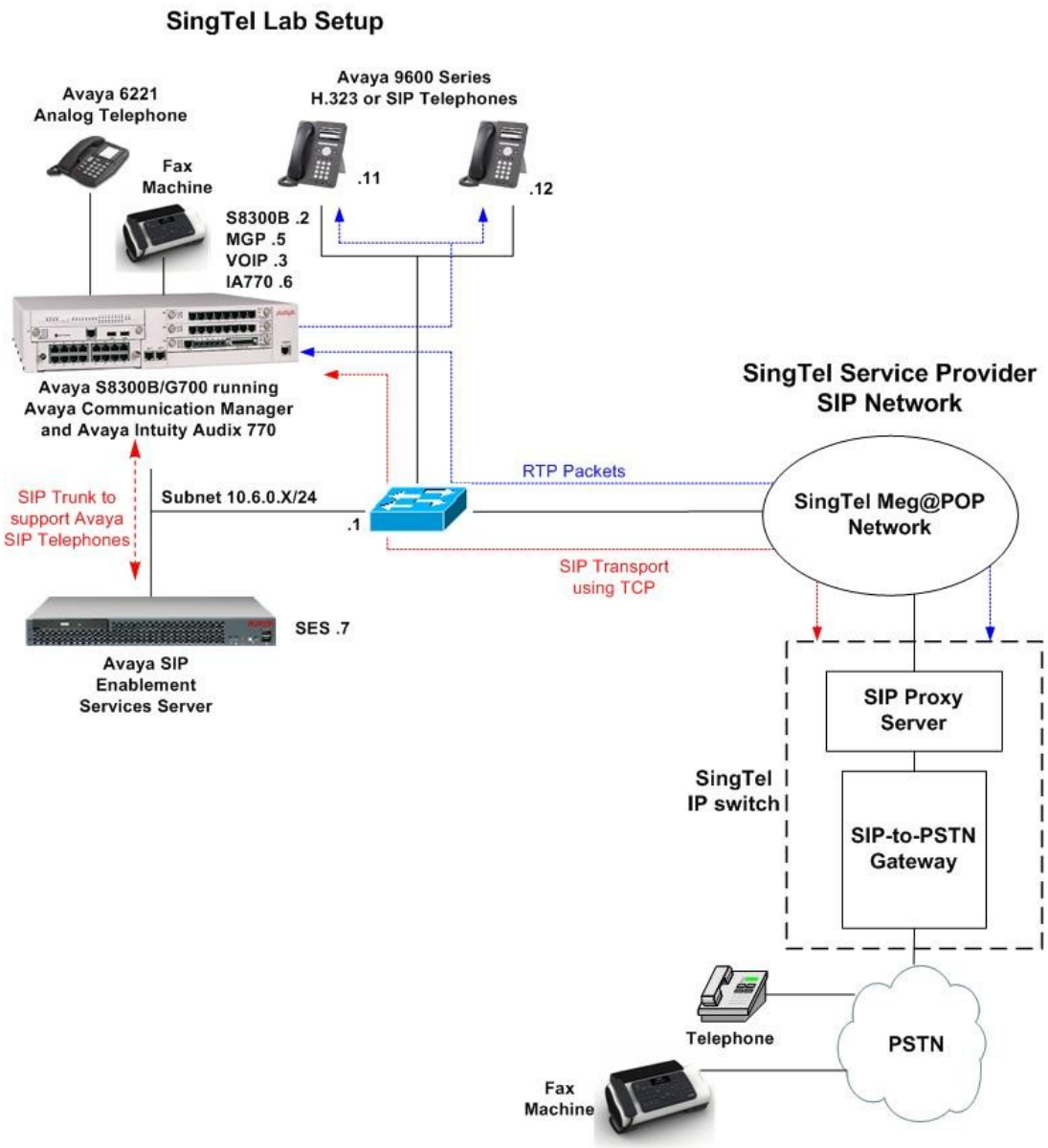


Figure 1: Avaya IP Telephony Network using SingTel Meg@POP SIP Trunking Service

2.1. 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 any telephone supported by Avaya Communication Manager.

1. A user on the PSTN dials a SingTel provided direct inward dialing (DID) number assigned to an Avaya Communication Manager telephone at the enterprise site. The PSTN 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 Communication Manager using SIP signaling messages sent over the converged access facility. Note that the assignment of the DID number and the address of the Avaya Communication Manager was previously configured in the SingTel IP Switch during the ordering and provisioning of the service.

3. Avaya Communication Manager terminates the call to the directly connected analog telephone as shown in **Figure 2** (**Step 3**). The same process occurs for calls to Avaya digital or H.323 IP telephones.

– or –

3a. Inbound calls destined for a SIP extension at the enterprise are routed from the Avaya Communication Manager to the Avaya SES over a SIP trunk.

4. Avaya SES transmits the appropriate SIP signaling to the SIP telephone (**Step 4**).

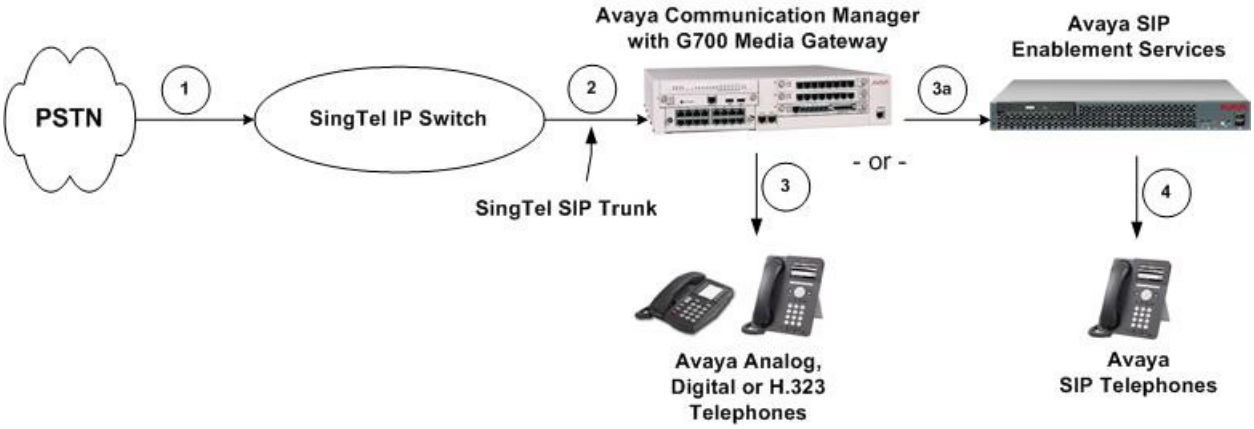


Figure 2: Incoming PSTN Calls to Avaya Communication Manager

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 via the SIP trunk to SingTel.

1. An Avaya H.323 IP, 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 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 SingTel SIP trunk and sends the SIP signaling messages to SingTel to route the call.

3. SingTel completes the call to the PSTN.

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

2.2. Dial Plan and Routing Administration for Fax

It is recommended that fax calls be set up using a G.711 codec for outgoing calls. This section outlines the overall administration strategy for Avaya Communication Manager to be implemented in the following sections of this document.

This strategy requires the following:

- Analog line ports must be specifically identified for fax (or modem use).
- Separate Avaya Communication Manager SIP trunk groups must be defined. One of the trunk groups must be created for fax / modem calls that use only the G.711 codec.
- The fax / modem stations must be assigned a Class of Restriction different from voice stations. The Facility Restriction Level (FRL) within this COR must have less calling privileges than voice stations.
- Outbound calls will use the Automatic Route Selection route patterns with FRL screening to use the correct trunk group.

3. 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 and Avaya IA 770 5.1.2 (Service Pack 01.2.416.4-17067)
Avaya G700 Media Gateway	MGP: 28.25.0 MM711 Analog: HW4/FW17 MM717 DCP: HW3/FW9
Avaya SIP Enablement Services on S8500B Media Server	5.1.2 (Service Pack SES-01.2.416.4-SP1)
Avaya 9620 IP Telephones (H.323)	3.0
Avaya 9620 IP Telephones (SIP)	2.0.5
Avaya Analog Telephone	-
G2 Fax Machine	-
SingTel VoIP Service Components	
Component	**Version**
Broadworks Softswitch	Release 13.0
Acme Packet Session Border Controller	Version C5.1.1
Cisco Router 2621xm	IOS Version 12.3

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 Servers and Media Gateway platforms running similar versions of Avaya Communication Manager and Avaya SIP Enablement Services on S8500 servers.

4. Configure Avaya Communication Manager

This section describes the steps for configuring the SIP trunks on Avaya Communication Manager. Direct SIP trunks are configured to allow Avaya Communication Manager to connect to the SingTel SIP Trunking Service. 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 messages are routed directly to the SingTel SIP Trunking Service.

A separate SIP trunk is established between Avaya Communication Manager and Avaya SIP Enablement Services (SES) server to allow SIP endpoint devices such as the Avaya 9600 Series SIP telephones to be configured as Outboard Proxy SIP (OPS) stations on the Avaya Communication Manager. OPS stations register with Avaya SES but have calling privileges and

features provided by Avaya Communication Manager. Note that the use of SIP endpoints is optional. The steps discussed in **Sections 4.2** and **Section 5** describing SIP endpoints administration may be omitted if SIP endpoints are not used.

The dial plan for the configuration described in these Application Notes consists of 8-digit dialing for local calls over the PSTN. However, Directory Assistance calls and International calls were also tested. Avaya Communication Manager routes all calls using Automatic Route Selection (ARS), except for intra-switch calls.

The configuration of Avaya Communication Manager was performed using the System Access Terminal (SAT). The general installation of the Avaya S8300B Server with G700 Media Gateway is presumed to have been previously completed. After the completion of the configuration, perform the **save translation** command to make the changes permanent.

4.1. SIP Trunk Configuration

Step 1: Confirm Necessary Optional Features
Login to the Avaya Communication Manager's SAT interface and confirm that sufficient Off PBX Telephone capacities for OPS are licensed. Use the **display system-parameters customer-options** command to determine these values. 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: V15                              Software Package: Standard
       Location: 2                              RFA System ID (SID): 1
       Platform: 7                              RFA Module ID (MID): 1

                                                             USED
                                Platform Maximum Ports: 900   65
                                      Maximum Stations: 450   13
                             Maximum XMOBILE Stations: 450   0
                     Maximum Off-PBX Telephones - EC500: 450   2
                     Maximum Off-PBX Telephones -   OPS: 450   5
                     Maximum Off-PBX Telephones - PBFMC: 450   0
                     Maximum Off-PBX Telephones - PVFMC: 450   0
                     Maximum Off-PBX Telephones - SCCAN: 450   0
```

Proceed to **Page 2** of **OPTIONAL FEATURES** form. Verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses.

Note: Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 450    12
           Maximum Concurrently Registered IP Stations: 450    0
             Maximum Administered Remote Office Trunks: 0      0
Maximum Concurrently Registered Remote Office Stations: 0      0
              Maximum Concurrently Registered IP eCons: 0      0
  Max Concur Registered Unauthenticated H.323 Stations: 450    0
                  Maximum Video Capable H.323 Stations: 450    0
                   Maximum Video Capable IP Softphones: 450    0
                       Maximum Administered SIP Trunks: 450    40
  Maximum Administered Ad-hoc Video Conferencing Ports: 0      0
   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: 300    0
```

Step 2: Assign Node Names
In the **IP Node-Names** form, assign the node name and IP address for the SingTel Session Border Controller (SBC). In this case **AcmeSD** and **200.10.10.10** are being used, respectively (Note: the actual IP address of the SingTel SBC is not used in these application notes). The SingTel SBC node name will be used throughout the other configuration screens of Avaya Communication Manager.

Note, this example shows the Avaya S8300B Server processor address **procr**, which is used as the SIP signaling interface in **Step 5**. If the Avaya IP Telephony solution utilizes an Avaya G650 Media Gateway, a CLAN can be used as the SIP signaling interface.

```
change node-names ip                                            Page   1 of   2
                                IP NODE NAMES
    Name                IP Address
AcmeSD                  200.10.10.10
default                 0.0.0.0
msgserver               10.6.0.6
procr                   10.6.0.2
singtelses              10.6.0.7
```

Step 3: Define IP Network Region
The **IP Network Region** form specifies the parameters used by the SIP trunk group connecting to the SingTel SIP Trunking Service. Configure the following:

- The **Authoritative Domain** field is configured to match the domain name configured on the SingTel SIP Trunking Service. This field is required for endpoints to call the public network. In this configuration, **200.10.10.10** is used as no DNS server is available. This address refers to the SingTel SBC IP address.
- By default, **IP-IP Direct Audio** (shuffling) for both **Intra-** and **Inter-region** is enabled to allow audio traffic to be sent directly between IP endpoints without using media

resources such as the TN2302AP IP Media Processor (MedPro) board or the VoIP engine in the Avaya G700 Media Gateway.

- 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, **Codec Set 2** is used. 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.

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.

```
change ip-network-region 2                                      Page   1 of  19
                              IP NETWORK REGION
  Region: 2
Location: 1       Authoritative Domain: 200.10.10.10
    Name: Singtel 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: 3329
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
```

On page 3, the **Source Region 2 (src rgn)** to **Destination Region 1 (dst rgn) Codec Set** is set as **2** to specify the codec set to be used for SIP Trunks to the enterprise site. In this case, the SIP trunk is assigned to a different IP network region than the G700 Media Gateway (IP Network Region 1).

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

                   Inter Network Region Connection Management

 src dst codec direct   WAN-BW-limits   Video      Intervening      Dyn
 rgn rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC IGAR AGL
 2   1    2     y    NoLimit                                            n
 2   2    2                                                                 all
 2   3    2     y    NoLimit                                            n
 2   4
 2   5
 2   6
 2   7    7     y    NoLimit                                            n
 2   8
 2   9
 2   10
 2   11
 2   12
 2   13
 2   14
 2   15
```

Step 4: Define IP Codecs

Bring up the **IP Codec Set** form using the ip-codec-set value specified in the **IP Network Region** form (**Step 3**) and enter the voice codec to be used for calls routed over the SingTel SIP Trunking Service. 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. During testing, the codecs were set as **G.729AB**, **G.729A**, **G.711MU** and **G.711A**. For the SingTel SIP Trunking Service, the preferred codec is G.729A. However, due to a difference in the way Avaya handles the G.729 MIME Type in the Session Description Protocol (SDP) parameters, Avaya recommends that the G.729AB codec is listed before the G.729A codec. **Media Encryption** must be set to **none** as SingTel SIP Trunking Service does not support media encryption.

```
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: 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:
```

On page 2 of the IP Codec Set form, set the **FAX Mode** to **t.38-standard** to support T.38 Fax calls.

```
change ip-codec-set 2                                          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
```

Step 5: Configure the Signaling Groups

For interoperability with SingTel, two signaling groups must be configured. The first signaling group is used for inbound and outbound calls. The second signaling group is used for inbound anonymous call 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** field of a signaling group, there are instances where 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. In order for inbound calls to be routed in a deterministic way, the second signaling group should be configured. This second signaling group can be thought of as a default signaling group. The steps below describe the configuration of these signaling groups.

Configure the first **Signaling Group** using the **add signaling-group s** command, where **s** is an available signaling group number as follows:

- Set the **Group Type** field to **sip**.
- Set the **Transport Method** field to **tcp**. As a result, the **Near-end Listen Port** and **Far-end Listen Port** fields are automatically set to **5060**.
- Set the **Near-end Node Name** field to **procr**. This node name maps to the IP address of the Avaya S8300B Server. Node names are defined using the **change node-names ip** command (see **Step 2**).
- Set the **Far-end Node Name** field to the node name of the SingTel SBC as defined in **Step 2**.
- For the **Far-end Network Region** field, enter the IP Network Region value configured in **Step 3**. When 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 (page 3 of IP Network Region form).
- Set the **Far-end Domain** field to the IP address of the SingTel SBC. This domain is specified in the Uniform Resource Identifier (URI) of the SIP "To" address in the

INVITE message. Configuring this field incorrectly may prevent calls from being successfully established to other SIP endpoints or to the PSTN.

- Set the **Direct IP-IP Audio Connections** field to **n** as shuffling is not supported over the SIP Trunk for SingTel.
- The **DTMF over IP** field should be set to the default value of **rtp-payload**. This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833.
- Set the **Enable Layer 3 Test** field to **y** to allow Avaya Communication Manager to use SIP OPTIONS messages to verify that they SIP trunk is in service.
- For the **Alternate Route Timer(sec)** field, a value of **18** is recommended to allow sufficient time for the call to be routed on the SingTel SIP Trunking Service.
- Use the default values for all other fields.

```
add signaling-group 2                                          Page   1 of   1
                              SIGNALING GROUP

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





   Near-end Node Name: procr                 Far-end Node Name: AcmeSD
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 2
       Far-end Domain: 200.10.10.10

                                            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? y
 Session Establishment Timer(min): 3            Alternate Route Timer(sec): 18
```

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 below.

```
add signaling-group 3                                          Page   1 of   1
                                SIGNALING GROUP

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





   Near-end Node Name: procr                  Far-end Node Name: AcmeSD
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 2
       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? y
 Session Establishment Timer(min): 3            Alternate Route Timer(sec): 18
```

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 and the other with the inbound signaling group.

Configure the first Trunk Group form using the **add trunk-group t** command, where **t** is an available trunk group number. In this case the trunk group number used is 2. On Page 1 of the TRUNK GROUP form:

- Set the **Group Type** field to **sip**.
- Enter a descriptive **Group Name**.
- Specify an available trunk access code (**TAC**).
- Set the **Service Type** field to **public-ntwrk**.
- Specify the first signaling group configured in **Step 5** to associate with this trunk group in the **Signaling Group** field.
- Specify the **Number of Members** supported by this SIP trunk group. Note that one trunk member is required for each call between Avaya Communication Manager and SingTel.

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

Group Number: 2                     Group Type: sip           CDR Reports: y
  Group Name: To SingTel                   COR: 1        TN: 1        TAC: #02
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                     Signaling Group: 2
                                                   Number of Members: 10
```

On Page 2 of the TRUNK GROUP form:

- Set the **Preferred Minimum Session Refresh Interval (sec)** field to **600** seconds. This field specifies the refresh INVITE Timer sent to the far-end.

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

TRUNK PARAMETERS

     Unicode Name? n
                                          Redirect On OPTIM Failure: 5000

            SCCAN? n                                Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 600
```

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.
- Set Show ANSWERED BY on Display field to n. This is the preferred setting for outgoing calls to the SingTel SIP Trunking Service.

```
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 Restricted Numbers? n
                                                 Replace Unavailable Numbers? n

 Show ANSWERED BY on Display? n
```

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 using the **change public-unknown numbering** command.

In this case, all stations with a 4-digit extension beginning with 1 should send the calling party number 65131xxx when an outbound call uses SIP trunk group number 2 (the outbound trunk group specified in **Step 6**). This calling party number will be sent to the far-end in the SIP "From" header.

```
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: 2
 4  1              2         6513              8           Maximum Entries: 240
 4  1              99                          4
```

Step 8: Automatic Route Selection for Outbound Calls
During compliance testing, the Automatic Route Selection (ARS) feature is used to route outbound calls via the SIP trunk to the SingTel SIP Trunking Service. 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
                                  Location:  all           Percent Full:    0

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

Use the **change feature-access-codes** command to specify **9** as the **Auto Route Selection (ARS) - Access Code** for outside dialing.

```
change feature-access-codes                                          Page   1 of   8
                              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: *99
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA: *91    All: *98    Deactivation: *97
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code:
                       Call Pickup Access Code: *96
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:
```

Next, use the **change ars analysis** command to configure the route pattern selection rule based upon the number dialed following the dialed ARS Access Code digit **9**. For compliance testing, when the dialed digit begins with **6**, the call is routed to **Route Pattern 1** which contains the trunk groups configured for the SingTel SIP Trunking Service. Note that further administration of ARS is beyond the scope of these Application Notes. For details, please refer to **[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
    001                   9    16    1        intl         n
    019                   9    16    1        intl         n
    1                     4    4     1        pubu         n
    100                   3    3     1        pubu         n
    1800                  11   11    1        pubu         n
    3                     8    8     1        pubu         n
    6                     8    8     1        pubu         n
    8                     8    8     1        pubu         n
    9                     3    3     1        pubu         n
    9                     8    8     1        pubu         n
```

Use the **change route-pattern** command to define the SIP trunk groups in the route pattern that ARS selects. Enter a descriptive name for the **Pattern Name** field. Set the **Grp No** field to the outgoing trunk group number created in **Step 6**. Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of **0** is the least restrictive level.

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

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 SingTel are manipulated as necessary to route calls to the proper extension on Avaya Communication Manager.

For this compliance testing, the incoming DID numbers provided by SingTel has a direct correlation to the internal extensions assigned within Avaya Communication Manager. Thus, the first four digits of incoming called number can be deleted and the remaining digits maps to the internal extension.

Use the **change inc-call-handling-trmt trunk-group t** command, where **t** is the trunk group to be configured. The Avaya dial plan has assigned 4 digits extensions from 1994 to 1999. Similarly, SingTel has assigned DID number range from 65131994 to 65131999. For extensions assigned a DID number from SingTel, enter **8** into the **Called Len,** and enter **4** into the **Del** fields, and the first four digits of the DID number into the **Called number** field. Leave the **Insert** field blank. **Called Number** entry in this case represents the common matching portion applicable to all incoming numbers. Thus, **6513** matches all numbers in the assigned DID range from SingTel.

```
change inc-call-handling-trmt trunk-group 2                    Page   1 of   3
                        INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called      Del Insert
 Feature        Len      Number
public-ntwrk    8   6513             4
```

If the customer's extension numbering does not align with the DID numbers, it may be necessary to define an entry for each DID number. Assuming Avaya Communication Manager has a 4-digit extension 1100, translating the DID number 65131998 to the extension would be done as shown below.

```
change inc-call-handling-trmt trunk-group 2                    Page   1 of   3
                        INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called      Del Insert
 Feature        Len      Number
public-ntwrk    8   65131998         8   1100
```

4.2. SIP Endpoint Configuration

This section describes the administration required to support SIP telephones. SIP telephones are optional and are not required to use the SingTel SIP Trunking Service.

Step 1: Assign Node Name for Avaya SES
In the **IP Node-Names** form, assign the node name and IP address for the Avaya SES. In this case **singtelses** and **10.6.0.7** are being used, respectively. The Avaya SES node name will be used throughout the other configuration screens of Avaya Communication Manager.

Note, this example shows the Avaya S8300B Server processor address **procr**, which is used as the SIP signaling interface in **Step 5**. If the Avaya IP Telephony solution utilizes an Avaya G650 Media Gateway, a CLAN can be used as the SIP signaling interface.

```
change node-names ip                                           Page   1 of   2
                                  IP NODE NAMES
    Name                IP Address
AcmeSD                  200.10.10.10
default                 0.0.0.0
msgserver               10.6.0.6
procr                   10.6.0.2
singtelses              10.6.0.7
```

Step 2: Define IP Network Region
The **IP Network Region** form specifies the parameters used by the SIP trunk group connecting to the Avaya SES. Configure the following:

- The **Authoritative Domain** field is configured to match the domain name configured on the Avaya SES in **Section 5 Step 2**. In this configuration, **avaya.com** is used.

- By default, **IP-IP Direct Audio** (shuffling) for both **Intra-** and **Inter-region** is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources such as the TN2302AP IP Media Processor (MedPro) board or the VoIP engine in the Avaya G700 Media Gateway.
- 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, **Codec Set 1** is used. This codec set will apply to calls located in this same IP Network Region (e.g. Avaya H.323 or SIP telephones) within the enterprise.

```
change ip-network-region 1                                      Page   1 of  19
                              IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name: Local
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                     IP Audio Hairpinning? n
   UDP Port Max: 3329
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
```

Step 3: Define IP Codecs

Bring up the **IP Codec Set** form using the ip-codec-set value specified in the **IP Network Region** form (**Step 2**) and enter the voice codec to be used for calls to the Avaya SIP telephones. During testing, the codecs were set as **G.711MU** and **G.711A**. **Media Encryption** is also set to **none** for this compliance testing.

```
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.711MU           n          2        20
 2: G.711A            n          2        20
 3:
 4:
 5:
 6:
 7:


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

On page 2 of the IP Codec Set form, set the **FAX Mode** to **off** as T.38 fax calls do not apply for the Avaya SIP Telephones.

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

                           IP Codec Set

                               Allow Direct-IP Multimedia? n



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

Step 4: Configure the Signaling Groups

This section describes the steps for administering a signaling group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SIP Enablement Services.

Enter the **add signaling-group s** command, where **s** is an available signaling group number and configure as follows:

- Set the **Group Type** field to **sip**.
- Set the **Transport Method** field to **tls**. As a result, the **Near-end Listen Port** and **Far-end Listen Port** fields are automatically set to **5061**.
- Set the **Near-end Node Name** field to **procr**. This node name maps to the IP address of the Avaya S8300B Server. Node names are defined using the **change node-names ip** command (see **Step 1**).

- Set the **Far-end Node Name** field to the node name of the Avaya SES as defined in **Step 1**.
- For the **Far-end Network Region** field, enter the IP Network Region value configured in **Step 1**.
- Set the **Far-end Domain** field to avaya.com. This should match the **SIP Domain** value configured on the Avaya SES in **Section 5 Step 2**.
- Set the **Direct IP-IP Audio Connections** field to **y** to allow shuffling between the Avaya SIP Telephones and other Avaya telephones.
- Use the default values for all other fields.

```
add signaling-group 1                                           Page   1 of   1
                                SIGNALING GROUP

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





   Near-end Node Name: procr                  Far-end Node Name: singtelses
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1
       Far-end Domain: avaya.com

                                            Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
                                                      IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 3             Alternate Route Timer(sec): 6
```

Step 5: Configure the Trunk Groups

This section describes the steps for administering a trunk group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

Enter the **add trunk-group t** command, where **t** is an available trunk group number. In this case the trunk group number used is 1. On Page 1 of the TRUNK GROUP form:

- Set the **Group Type** field to **sip**.
- Enter a descriptive **Group Name**.
- Specify an available trunk access code (**TAC**).
- Set the **Service Type** field to **tie**.
- Specify the first signaling group configured in **Step 4** to associate with this trunk group in the **Signaling Group** field.
- Specify the **Number of Members** supported by this SIP trunk group. Note that each call between two SIP endpoints (whether internal or external) requires two SIP 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: To Avaya SES                 COR: 1         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
```

Step 6: **Assign a Station**

The first step in adding an OPS station for Avaya SIP telephones registered with Avaya SES is to assign a station using the **add station e** command, where **e** is an available extension valid in the dial plan. On Page 1 of the STATION form, configure the following:

- Leave the station **Type** at the default "6408D+" value. (Note this is the Avaya recommended best practice that will prevent an alarm warning which occurs when an IP phone model is entered).
- Enter "X" in the **Port** field to indicate station administration without port hardware.
- Enter a **Name** for the station.
- The **Security Code** is left blank for OPS stations.

The remaining fields are configured as per normal station administration, which 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 1995                                                 Page   1 of   5
                                     STATION

Extension: 1995                         Lock Messages? n               BCC: 0
     Type: 6408D+                        Security Code:                 TN: 1
     Port: X                           Coverage Path 1:                COR: 1
     Name: John Doe                    Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 2          Personalized Ringing Pattern: 1
             Data Module? n                     Message Lamp Ext: 1995
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english

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

On Page 2 of the STATION form, set the **Restrict Last Appearance** field to **n** on phones that have 3 or fewer call appearances in order to maintain proper SIP conference and transfer operation. Setting the **Restrict Last Appearance** field 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 1995                                                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

        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
        Multimedia Mode: basic                     Audible Message Waiting? n
    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: 1995                        IP Audio Hairpinning? N
```

On Page 4 of the STATION form, configure at least 3 call appearances under the **Button Assignments** section for the Avaya SIP telephone.

```
add station 1995                                                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:
 2: call-appr                          6:
 3: call-appr                          7:
 4:                                    8:
```

Step 7: **Configure Off-PBX Station Mapping**
Configure the **Off-PBX Telephone** form so that calls destined for a SIP telephone at the enterprise site are routed to Avaya SES, which will then route the call to the SIP telephone.

On the STATIONS WITH OFF-PBX TELEPHONE INTEGRATION form, configure the following:

- 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 SES also match the extensions of the corresponding administration without hardware (AWOH) stations on Avaya Communication Manager.
- Set the **Trunk Selection** field to **1**, which is the trunk group number of the SIP trunk group between Avaya Communication Manager and Avaya SES configured in **Step 5**.
- Set the **Config Set** value to **1**. In these Application Notes, Configuration Set 1 uses the default values of the Configuration Set form.

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

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 1995              OPS          -     1995                 1          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 **3**, which corresponds to the number of call appearances configured on the STATION form. Accept the default values for the other fields.

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

 Station       Call       Mapping      Calls       Bridged        Location
 Extension     Limit      Mode         Allowed     Calls
 1995           3       both           all         both
```

Repeat **Steps 6** and **7** as necessary for each Avaya SIP telephone to be added.

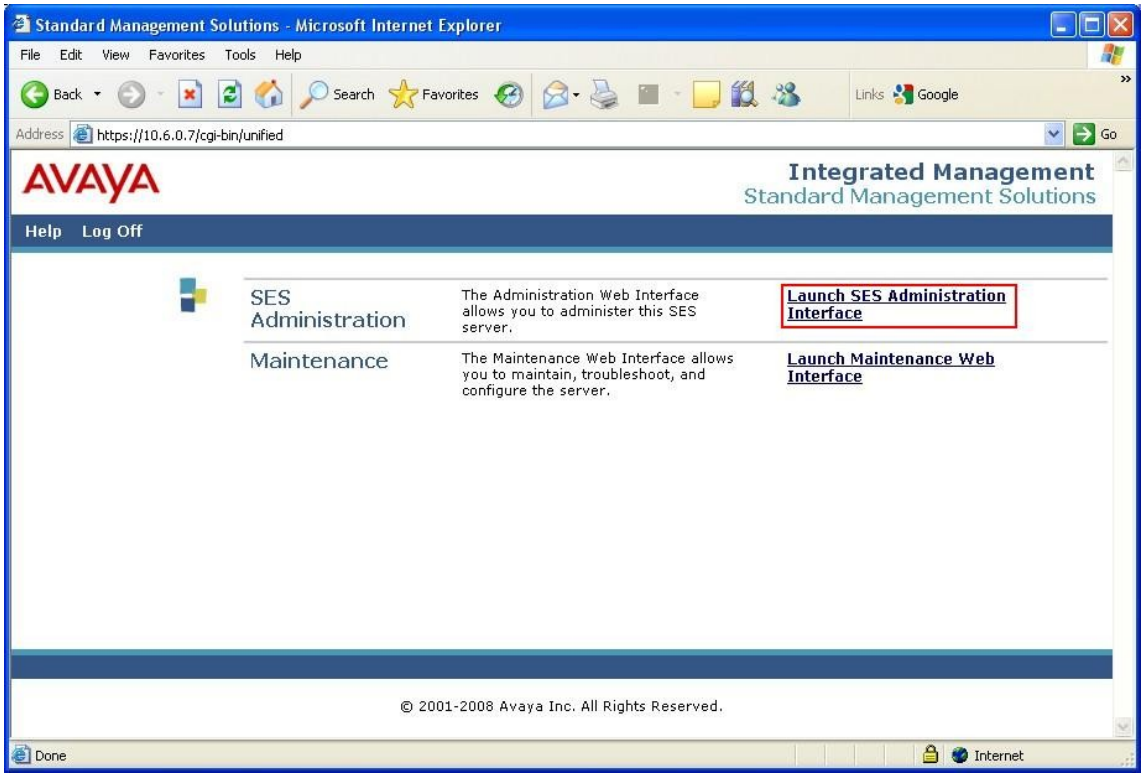
5. Configure Avaya SIP Enablement Services

This section covers the administration of Avaya SES. The configuration of the Avaya SES is optional and is only required to support SIP endpoints such as the Avaya 9600 Series SIP Telephones. The Avaya SES is configured via an Internet browser using the Administration web interface. It is assumed that the Avaya SES software and the license file have already been installed on the Avaya SES. 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 Avaya SES parameters. For additional information on these installation tasks, refer to **[4]**.

Step 1: Access Avaya SES

Access the Avaya 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 SES server.

Log in with the appropriate credentials and then click **Launch SES Administration Interface** from the main screen.

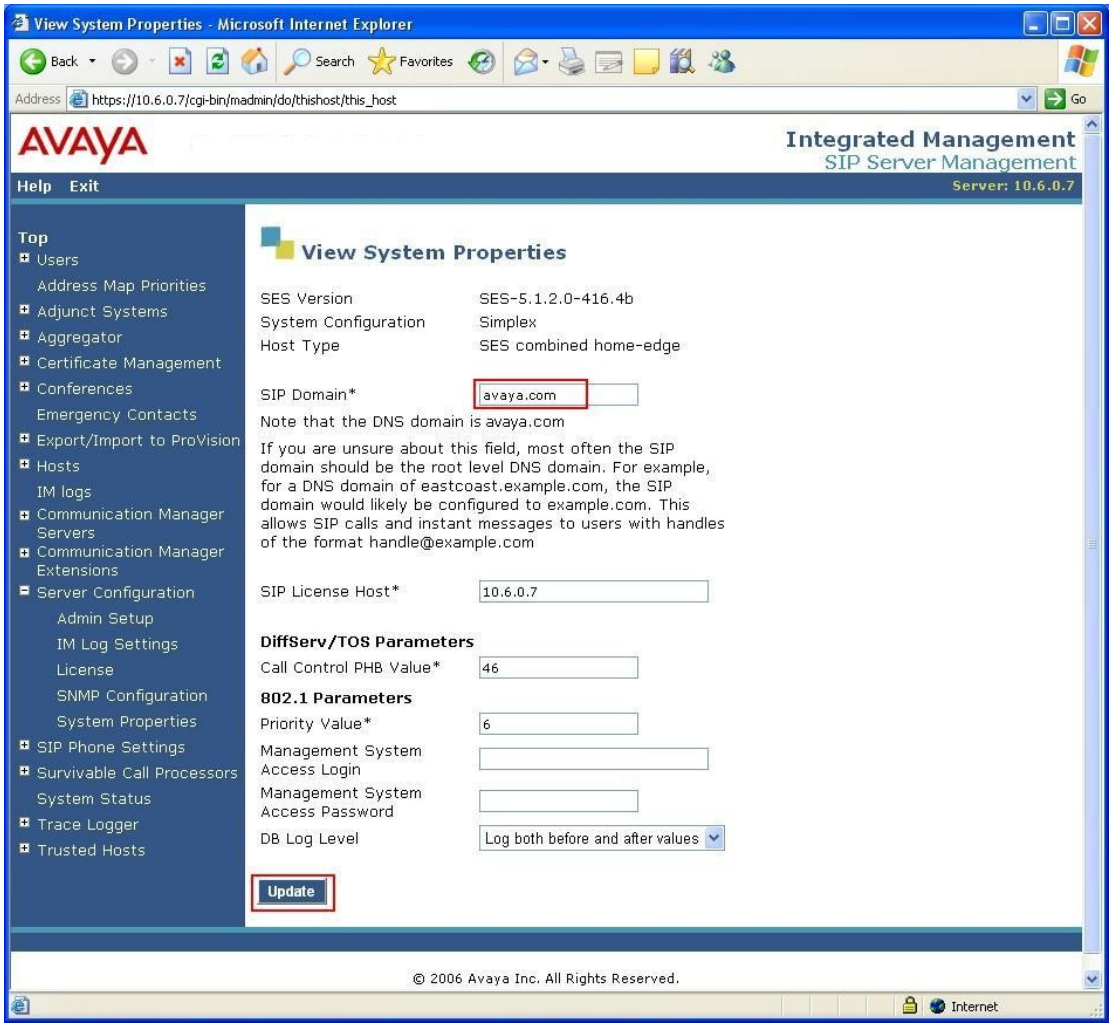


Step 2: Define System Properties

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

In the View System Properties screen:

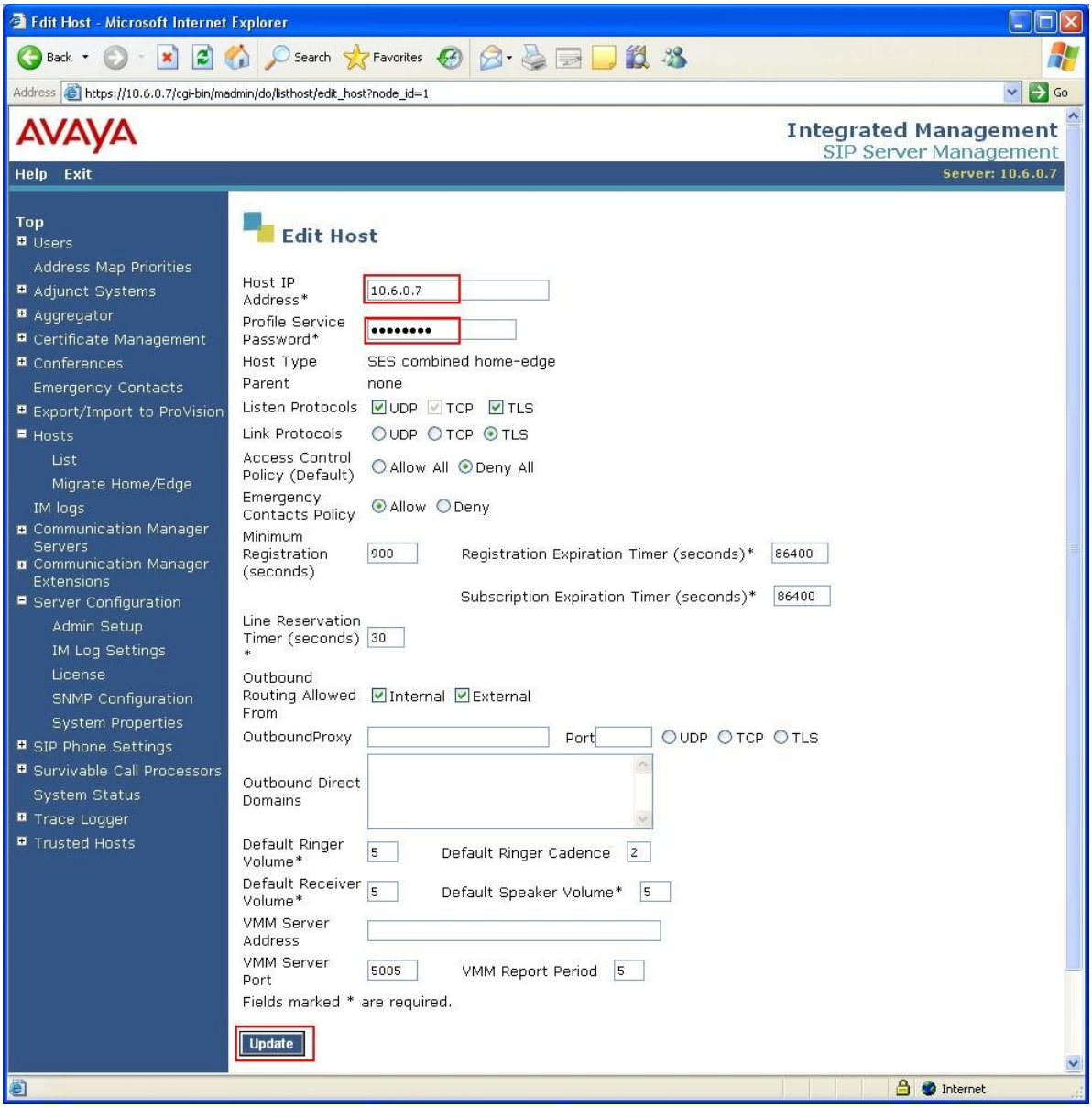
- Enter the **SIP Domain** name assigned to Avaya SES. In this configuration, the SIP domain is **avaya.com**.
- Enter the **License Host** field. This is the host name, the fully qualified domain name, or the IP address of the Avaya SES server that is running the WebLM application and has the associated license file installed.
- After configuring the **System Properties** screen, click **Update**.



Step 3: Enter Avaya SES Host Information
Create a host entry for the Avaya SES. The following example shows the **Edit Host** screen since the host had already been added to the system.

The **Edit Host** screen is accessed by expanding the **Hosts** option in the left pane and then clicking **List**. From the List Hosts screen, click **Edit** for the host entry that appears on the subsequent page that is displayed (not shown).

- Enter the IP address 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.
- Click **Update**.



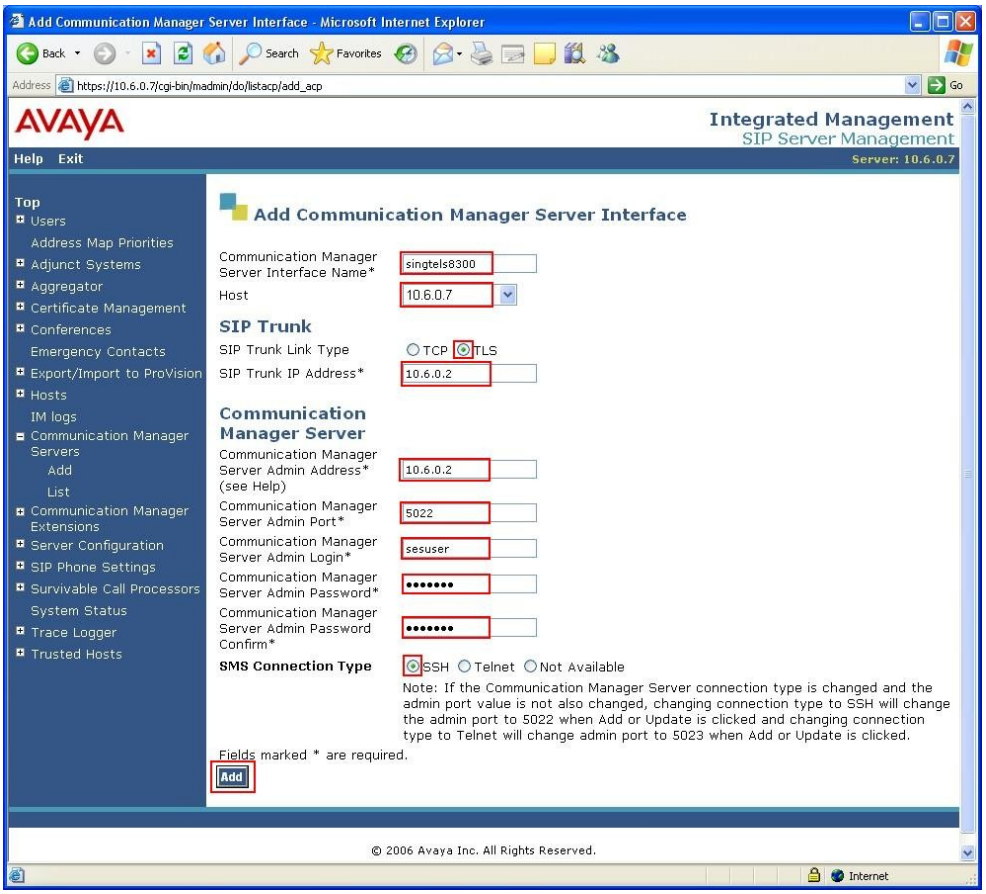
Step 4: **Add Communication Manager Server**

Under the **Communication Manager 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 **Section 4.2**.

In the **Add Communication Manager Server Interface** screen, enter the following information:

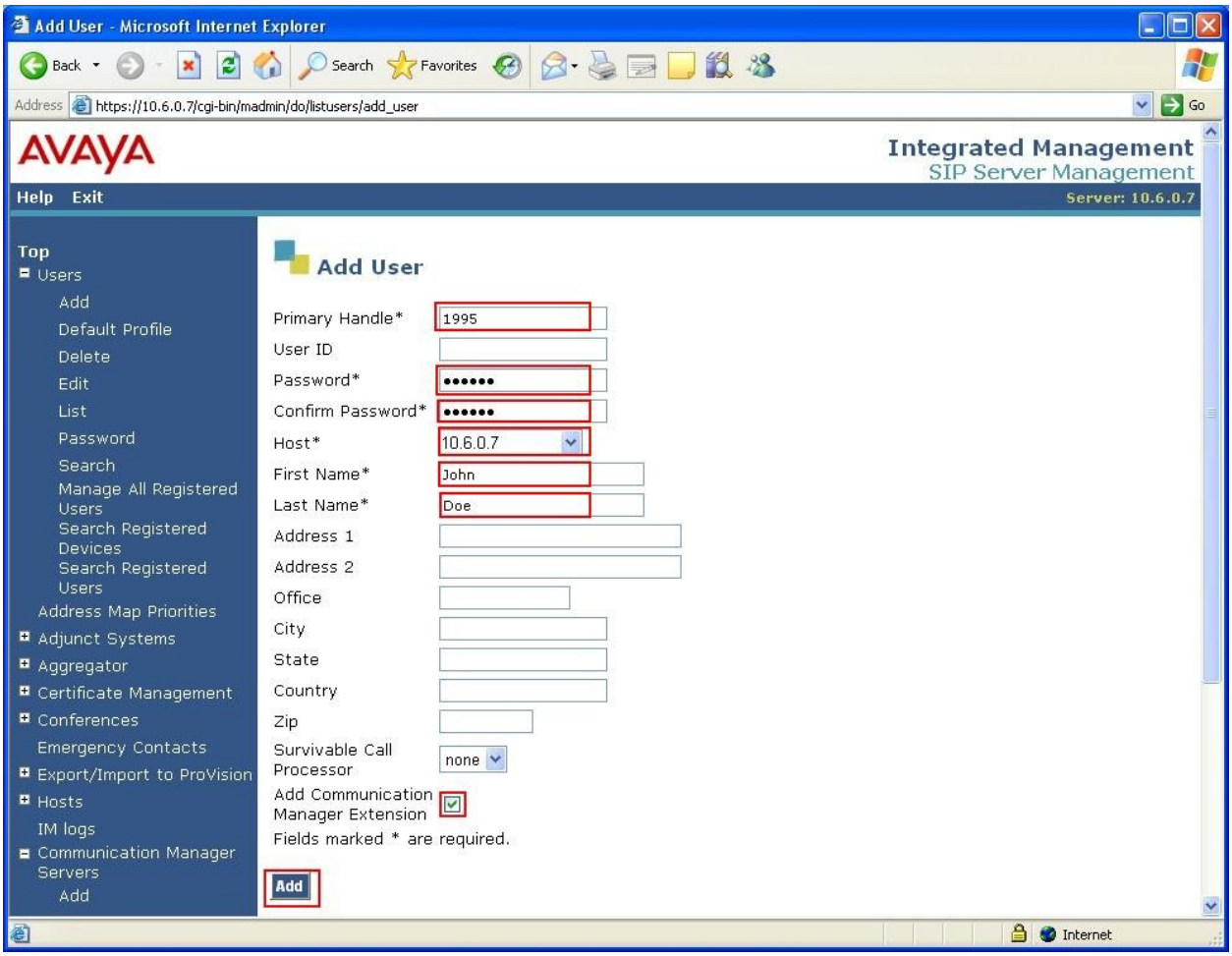
- A descriptive name in the **Communication Manager Server Interface Name** field (e.g., **singtels8300**).
- Select **TLS** (Transport Link Security) for the **SIP Trunk Link Type**. TLS provides encryption at the transport layer.
- Enter the IP address of the Avaya S8300B Server in the **SIP Trunk IP Address** field. (Note: This may be the IP address of the CLAN board in larger Avaya Communication Manager configurations such as the Avaya S8720 Media Servers using an Avaya G650 Media Gateway.)
- Enter the IP address of the Avaya S8300B Server for **Communication Manager Server Admin Address**.
- Leave **Communication Manager Server Admin Port** at the default value of **5022**.
- Enter the **Communication Manager Server Admin Login** provided for Avaya SES to access Avaya Communication Manager and the **Communication Manager Server Admin Password**. For this compliance testing, a new administrator login sesuser was created using the Maintenance Web Pages of the Avaya Communication Manager.
- Select **SSH** for SMS Connection Type.

After completing the **Add Communication Manager Server Interface** screen, click **Add**.



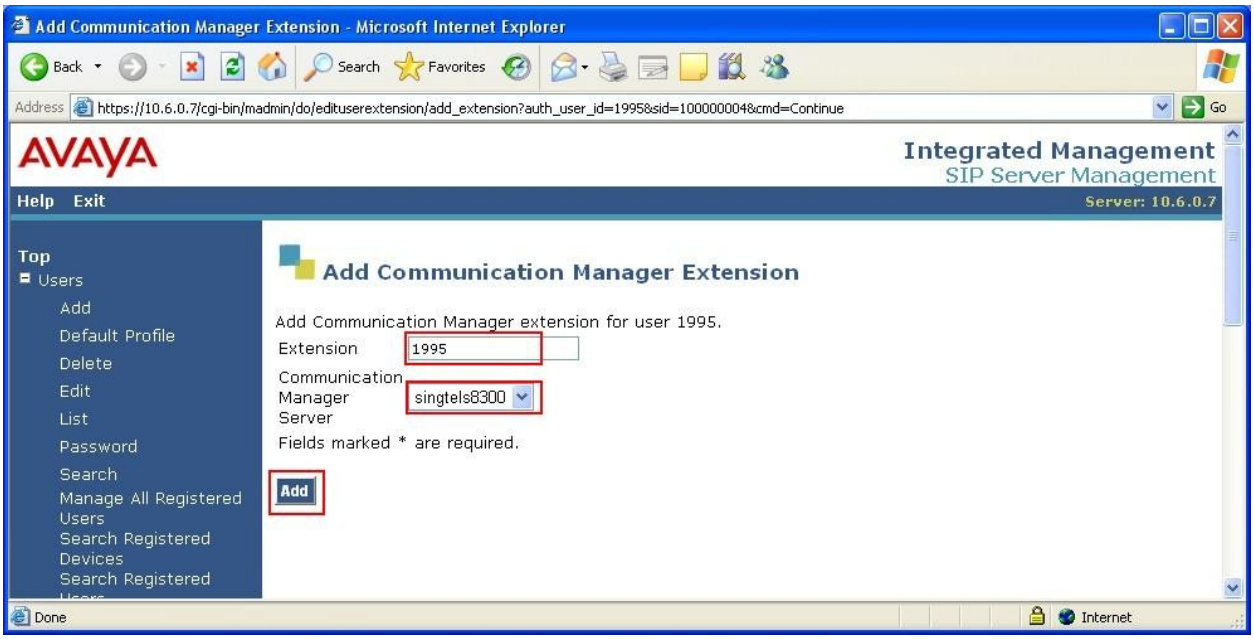
Step 4: **Add a SIP User**
In Avaya SES administration, expand the **Users** link in the left navigation bar and click on **Add**. In the **Add User** screen, configure the following:

- 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 (**10.6.0.7**) for this user.
- Enter the **First Name** and **Last Name** of the user.
- To associate a Communication Manager extension with this user, select the **Add Communication Manager 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 (not shown).



Step 5: Specify Corresponding Avaya Communication Manager Extension
The SIP telephone primary handle must now be associated with the corresponding extension on Avaya Communication Manager. In the **Add Communication Manager Extension** screen, configure the following:

- Enter the **Extension** configured on the Avaya Communication Manager for the OPS extension previously defined in Section 4.2. Usually, the Communication Manager extension and the user extension are the same (recommended) but it is not required to be.
- Select the **Communication Manager Server** assigned to this extension.
- Click **Add**.



Repeat **Steps 4** and **5** as necessary for each new SIP user.

6. SingTel SIP Trunking Service Configuration

In order to use SingTel SIP Trunking Service on the Meg@POP IP VPN Network, a customer must order the service from SingTel. For further information on SingTel Meg@POP as well as its network and access services, contact a SingTel Account Manager or call 1800-763-1111 (local toll-free).

7. General Test Approach and Test Results

Interoperability testing of the sample configuration was completed with successful results. The following items described below were observed.

Item	Issue Observed	Discussion / Workaround
Call Forwarding All/Busy or Don't Answer	PSTN incoming call forwarded to another PSTN outgoing number fails. This issue does not happen to forwarding to a phone number that is registered directly with Broadworks.	The cause was due to differences adopted for reading the call forwarding information. Avaya Communication Manager will implement SIP Diversion Header, which is used by Broadworks, in Release 5.2.
Outgoing T.38 Fax to the PSTN	Only G.711 Codec was supported for outgoing fax.	Configure the fax machine to appropriately select the SIP Trunk that is configured to use G.711 only. See **Section 2.2** on the strategy.

8. Verification Steps

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

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 required to verify that proper routing of the SIP messages and the SIP protocol timers has been satisfied.
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.

9. Conclusion

These Application Notes describe the configuration steps required to connect customers using Avaya Communication Manager to the SingTel SIP Trunking Service on the SingTel Meg@POP IP VPN Network. SIP trunking uses the Session Initiation Protocol to connect private company networks to the public telephone network via converged IP access. It provides businesses a flexible, cost-saving alternative to current TDM-based telephony trunk lines.

10. Additional 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*, Release 5.0, Issue 4.0, January 2008, Document Number 03-300509.
[2] *Feature Description and Implementation for Avaya Communication Manager,* Issue 6, January 2008, Document Number 555-245-205.
[3] *Avaya Extension to Cellular User Guide Release 5.0*, January 2008, Issue 11, Document Number 210-100-700.
[4] *Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services*, Issue 6.0, June 2008, Document Number 03-600768.
[5] *SIP Support in Avaya Communication Manager Running on Avaya S8XXX Servers*, Issue 8, January 2008, Document Number 555-245-206.

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 Communication Manager:

```
INVITE sip:65131994@avaya.com:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 200.10.10.10:5060;branch=z9hG4bKevcfn32028v1ae4b5480.1
From: "62738056"<sip:62738056@200.10.10.10;user=phone>;tag=922961227-1239867371713-
To: <sip:65131994@avaya.com>
Call-ID: BW153611713160409-601838693@202.163.63.65
CSeq: 384653409 INVITE
Contact: <sip:62738056@200.10.10.10:5060;transport=tcp>
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Supported:
Accept: multipart/mixed,application/gtd,application/media_control+xml,application/sdp
Max-Forwards: 69
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 371

v=0
o=BroadWorks 3424455 1 IN IP4 200.10.10.10
s=-
c=IN IP4 200.10.10.10
t=0 0
m=audio 35468 RTP/AVP 18 0 8 101 117
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:117 X-NSE/8000
a=fmtp:117 192-194,200-202
a=X-sqn:0
a=X-cap: 1 audio RTP/AVP 117
a=X-cpar: a=rtpmap:117 X-NSE/8000
a=X-cpar: a=fmtp:117 192-194,200-202
a=X-cap: 2 image udptl t38
```

Sample SIP INVITE Message from Avaya Communication Manager to SingTel:

```
INVITE sip:61111111@200.10.10.10 SIP/2.0
From: "9620 H323" <sip:65131994@avaya.com>;tag=026661434de1db049f98a2600
To: "61111111" <sip:61111111@200.10.10.10>
Call-ID: 026661434de1dc049f98a2600
CSeq: 1 INVITE
Max-Forwards: 69
Route: <sip:200.10.10.10;lr;phase=terminating;transport=tcp>
Record-Route: <sip:10.6.0.2;lr;transport=tcp>
Via: SIP/2.0/TCP 10.6.0.2;branch=z9hG4bK026661434de1dd049f98a2600
User-Agent: Avaya CM/R015x.01.2.416.4
Supported: 100rel, timer, replaces, join, histinfo
Allow: INVITE, CANCEL, BYE, ACK, PRACK, SUBSCRIBE, NOTIFY, REFER, OPTIONS, INFO, PUBLISH
Contact: "9620 H323" <sip:65131994@10.6.0.2;transport=tcp>
Session-Expires: 3600;refresher=uac
Min-SE: 3600
P-Asserted-Identity: "9620 H323" <sip:65131994@avaya.com>
Accept-Language: en
Content-Type: application/sdp
History-Info: <sip:61111111@200.10.10.10>;index=1
History-Info: "61111111" <sip:61111111@200.10.10.10>;index=1.1
Alert-Info: <cid:internal@200.10.10.10>;avaya-cm-alert-type=internal
Content-Length: 206

v=0
o=- 1 1 IN IP4 10.6.0.2
s=-
c=IN IP4 10.6.0.3
b=AS:64
t=0 0
m=audio 12006 RTP/AVP 18 8 0 101
a=rtpmap:18 G729/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
```

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.