Avaya Solution & Interoperability Test Lab

Application Notes for Level 3 Enterprise IP Trunking Service with an Avaya IP Telephony Network - Issue 1.0

Abstract

These Application Notes describe the steps for configuring SIP trunking between the Level 3 VoIP Network and an Avaya IP Telephony Network consisting of Avaya SIP Enablement Services and Avaya Communication Manager. Avaya IP, digital and analog endpoints were used to originate and terminate calls. Enterprise customers with an Avaya SIP-based network can communicate with the Level 3 VoIP Network over the Internet using Session Initiation Protocol (SIP) and access the PSTN by subscribing to the *Level 3 Enterprise IP Trunking* service. This solution allows enterprise customers with a converged network to reduce long distance and interconnection costs.

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 Level 3 VoIP Network and an Avaya IP Telephony Network consisting of Avaya SIP Enablement Services and Avaya Communication Manager. Avaya IP, digital and analog endpoints were used to originate and terminate calls. Enterprise customers with an Avaya SIP-based network can communicate with the Level 3 VoIP Network over the Internet using Session Initiation Protocol (SIP) and access the PSTN by subscribing to the *Level 3 Enterprise IP Trunking* service. This solution allows enterprise customers with a converged network to reduce long distance and interconnection costs.

SIP is a signaling protocol designed to provide a common framework for session establishment, modification, and termination for supporting multimedia communications including voice and video. In converged communications, SIP acts as a trunking protocol, enabling the direct interconnection of independent systems with a SIP network interface.

Figure 1 illustrates an enterprise site with an Avaya SIP-based network, including Avaya SIP Enablement Services, a pair of Avaya S8710 Servers with a G650 Media Gateway[1] running Avaya Communication Manager, and Avaya IP, digital, and analog endpoints. The enterprise site is connected to the Level 3 VoIP Network over the Internet and communicates using SIP. The Level 3 VoIP Network consists of Acme Packet Session Border Controller, Broadsoft BroadWorks VoIP Applications Platform, Sonus Networks Network Border Switch (NBS), and Sonus Networks GSX Gateways. The Acme Packet Session Border Controller exchanges SIP signaling messages with Avaya SIP Enablement Services.

[1] This solution is compatible with other Avaya Server and Media Gateway platforms running Avaya Communication Manager.

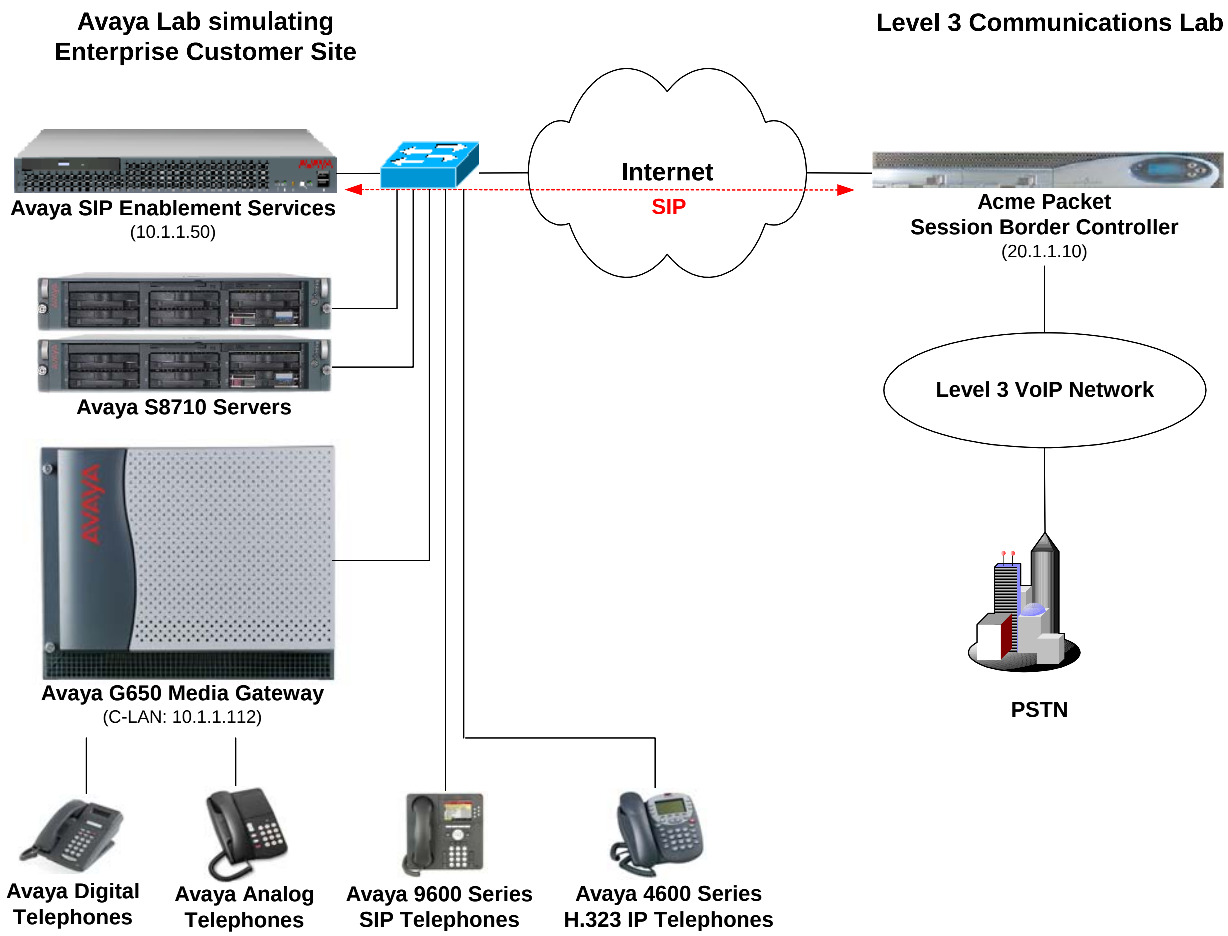

Figure 1: Avaya IP Telephony Network connected to the Level 3 VoIP Network

1.1. SIP Call Flows

To better understand how calls are routed between the PSTN and the enterprise site shown in **Figure 1**, two call flows are described in this section. The first call scenario is a PSTN call to the enterprise site and the second call scenario is an outbound call from the enterprise site to the PSTN. In both cases, the call transits the Level 3 VoIP Network. **Figure 2** illustrates the call flow for a call originated from the PSTN and terminated at the enterprise site.

1. A user on the PSTN dials a DID number assigned to an Avaya SIP telephone at the enterprise site. The enterprise site subscribes to the Level 3 Enterprise IP Trunking service so the call is routed through the Level 3 VoIP network.
2. Based on the DID number, Level 3 routes the call to the enterprise site via SIP trunking. Level 3 sends SIP signaling messages to Avaya SIP Enablement Services at the enterprise site. See the Appendix A for an example of a SIP INVITE message sent by Level 3.
3. Avaya SIP Enablement Services routes the call to the Avaya S8710 Server running Avaya Communication Manager over a SIP trunk.
4. Since the call is destined for an Avaya SIP telephone, Avaya Communication Manager routes the call back to Avaya SIP Enablement Services over a SIP trunk. If the destination of the call was an H.323, digital or analog endpoint, Avaya Communication Manager would terminate the call directly to the endpoint and steps 4 and 5 would not be required.
5. Avaya SIP Enablement Services terminates the call to the Avaya SIP telephone.

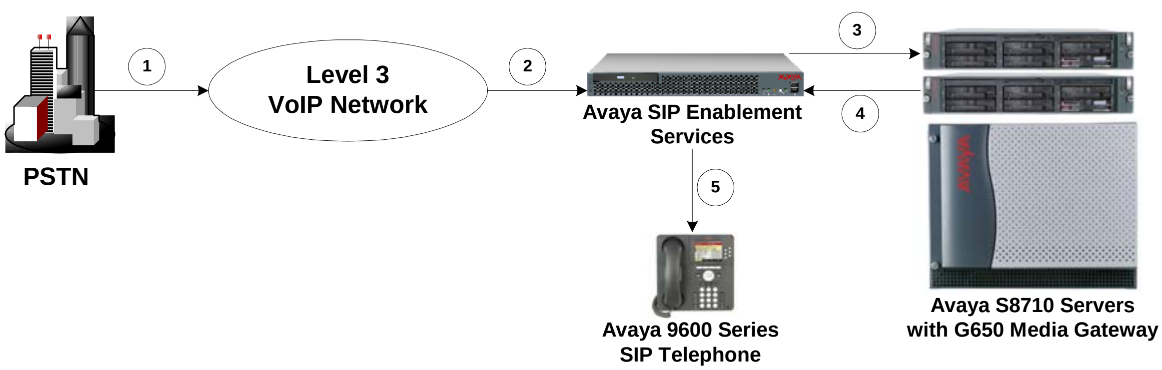

Figure 2: PSTN Call to the Avaya SIP Network

Figure 3 illustrates the call flow for an outgoing call from an Avaya SIP telephone on the Avaya SIP network at the enterprise site to the PSTN.

1. An Avaya SIP telephone originates a call to a user on the PSTN. The call request is delivered to Avaya SIP Enablement Services. If the originator were an H.323, digital or analog endpoint, the call request would be sent to Avaya SIP Enablement Services from Avaya Communication Manager.
2. Avaya SIP Enablement Services routes the call over the SIP trunk to the Avaya S8710 Server running Avaya Communication Manager for origination services. This allows Avaya Communication Manager to apply the appropriate call restrictions to the endpoint, handle call routing, and track the status of the SIP telephone, which is an off-PBX station.
3. After applying the origination services, Avaya Communication Manager routes the call back to Avaya SIP Enablement Services over a SIP trunk.
4. Avaya SIP Enablement Services routes the call to the Level 3 VoIP Network. See the Appendix A for an example of a SIP INVITE message sent by the Avaya SIP-based network.
5. Level 3 routes the call to the PSTN.

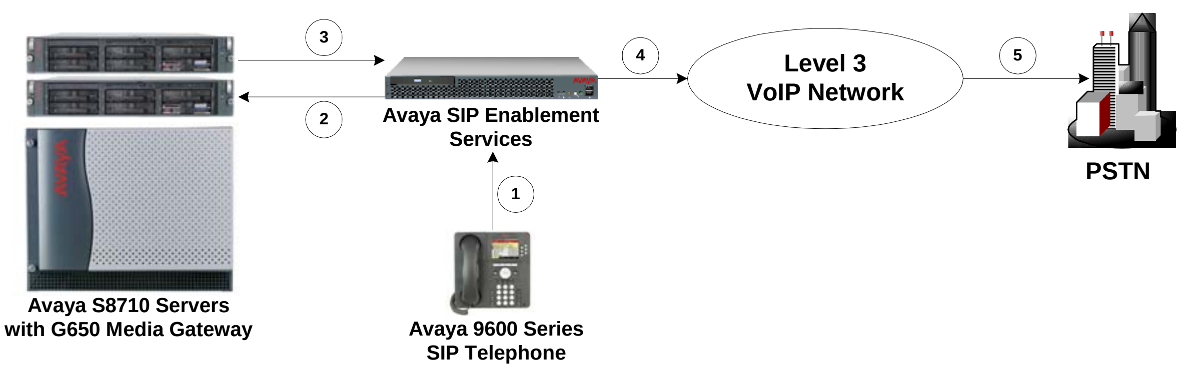

Figure 3: Avaya SIP Call to the PSTN

2. Equipment and Software Validated

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

Hardware Component	Version
Avaya S8710 Servers with a G650 Media Gateway	Communication Manager 5.0 (R015x.00.0.825.4)
Avaya SIP Enablement Services	5.0 (SES-5.0.0.0-825-31)
Avaya 4600 Series IP Telephone	2.8 (H.323)
Avaya 9600 Series IP Telephones	2.0.4 (SIP)
Avaya Digital Telephones	--
Avaya Analog Telephones	--
Acme Packet Session Border Controller	C5.0.0.0 Patch 2
Sonus Networks Network Border Switch (NBS) and GSX Gateway	6.4
Broadsoft BroadWorks VoIP Applications Platform	R14 SP2

3. Configure Avaya Communication Manager

This section describes the steps for configuring a SIP trunk and off-PBX stations (OPS) on Avaya Communication Manager. The SIP trunk is established between Avaya Communication Manager and Avaya SIP Enablement Services. An off-PBX station (OPS) is configured for each Avaya SIP telephone registered with Avaya SIP Enablement Services. Refer to [2] for additional information on configuring an off-PBX station. All incoming calls from Level 3 are received by Avaya SIP Enablement Services and routed to Avaya Communication Manager over a SIP trunk for termination services. All outbound calls to the PSTN are routed through Avaya Communication Manager for origination services. Avaya Communication Manager then routes the call to Avaya SIP Enablement Services, which in turn routes the call to the PSTN through the Level 3 VoIP network. Note that Avaya SIP Enablement Services provides the SIP interface to the Level 3 VoIP Network.

The dial plan for the configuration described in these Application Notes consisted of 10-digit dialing for local and long-distance calls over the PSTN. In addition, Directory Assistance calls (411) and International calls (011 Country Code) were also supported. Avaya Communication Manager routed all calls using Auto Route Selection (ARS), except for intra-switch calls. Configuring ARS is beyond the scope of these Application Notes and the reader should refer to [1] for additional information.

Avaya Communication Manager configuration was performed using the System Access Terminal (SAT). The IP network parameters of the Avaya S8710 Servers were configured via the Maintenance web interface using an Internet browser (not shown here). Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks features are enabled on the **System-Parameters Customer-Options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative. On Page 1, verify that the number of OPS stations allowed in the system is sufficient.

```
display system-parameters customer-options                      Page   1 of  10
                                OPTIONAL FEATURES
     G3 Version: V15                              Software Package: Standard
       Location: 1                             RFA System ID (SID): 1
       Platform: 6                             RFA Module ID (MID): 1
                                                               USED
                                Platform Maximum Ports: 44000 253
                                      Maximum Stations: 36000 60
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 10    2
                    Maximum Off-PBX Telephones -   OPS: 36000 20
                    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 of the **System-Parameters Customer-Options** form, verify that the number of SIP trunks supported by the system is sufficient.

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

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 100   40
           Maximum Concurrently Registered IP Stations: 12000 2
             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: 5     0
                  Maximum Video Capable H.323 Stations: 10    0
                   Maximum Video Capable IP Softphones: 10    0
                       Maximum Administered SIP Trunks: 200   130
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 1     0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 50    0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   1
          Maximum TN2602 Boards with 320 VoIP Channels: 128   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

On the **System-Parameters Features** form, set the **Trunk-to-Trunk Transfer** field to *all* to allow calls to be transferred from the enterprise site to an endpoint on the PSTN. Otherwise, leave the field set to *none*. The SIP call flows described in Section 1.1 did not require trunk-to-trunk transfer to be enabled.

```
change system-parameters features                               Page   1 of  17
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y
                             Music/Tone on Hold: none
              Music (or Silence) on Transferred Trunk Calls? no
                       DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n

             Abbreviated Dial Programming by Assigned Lists? n
       Auto Abbreviated/Delayed Transition Interval (rings): 2
                     Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? n
```

Figure 6: System-Parameters Features Form

In the **IP Node Names** form, assign an IP address and host name for the C-LAN board in the Avaya G650 Media Gateway and for Avaya SIP Enablement Services at the enterprise site. The host names will be used throughout the other configuration screens of Avaya Communication Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name               IP Address
clan                 10.1.1.112
default              0.0.0.0
medpro-hw11          10.1.1.116
ses-he               10.1.1.50
( 4 of 19   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 7: IP Nodes Names Form

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Avaya SIP Enablement Services. In this configuration, the domain name is *east.devcon.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G650 Media Gateway. In addition, DTMF transmission using RFC 2833 (described later) is also required for shuffling among IP devices as shown in **Figure 11**. The **IP Network Region** form also specifies the **IP Codec Set** to be used for local calls and calls routed over the SIP trunk to Avaya SIP Enablement Services. This codec set is used when its corresponding network region (i.e., IP Network Region ‘1’) is specified in the SIP signaling group as shown in **Figure 11**.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: east.devcon.com
    Name: Default System All
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? y
   UDP Port Max: 65531
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
```

Figure 8: IP Network Region Form

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set ‘1’ was specified in IP Network Region ‘1’ shown in **Figure 8**. The default settings of the **IP Codec Set** form are shown below. However, the **IP Codec Set** form may specify multiple codecs, including G.711 and G.729, to allow the codec for the call to be negotiated during call establishment. However, G.729B is not supported by Level 3 for IP-to-PSTN calls.

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

                          IP Codec Set

    Codec Set: 1

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

Figure 9: IP Codec Set – Page 1

To enable Fax T.38, set the Fax mode on Page 2 of the IP codec set form to *t.38-standard*.

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n

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

Figure 10: IP Codec Set – Page 2

Prior to configuring a SIP trunk group for communication with Avaya SIP Enablement Services, a SIP signaling group must be configured. This signaling group is used for outgoing calls to the PSTN. Configure the Signaling Group form shown in **Figure 11** as follows:

- Set the **Group Type** field to *sip*.
- The **Transport Method** field will default to *tls* (Transport Layer Security).
- Specify the C-LAN board in the G650 Media Gateway and the Avaya SIP Enablement Services Server as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form shown in **Figure 7**.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field. Although the same network region (Network Region 1) was used for local and PSTN calls in this configuration, a different network region could have been specified in this field.
- Enter the domain name of Avaya SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is *east.devcon.com*. This domain is specified in the Uniform Resource Identifier (URI) of the "SIP To Address" in the INVITE message. Mis-configuring this field may prevent calls from being successfully established to other SIP endpoints or to the PSTN.
- If calls to/from SIP endpoints are to be shuffled, then the **Direct IP-IP Audio Connections** field must be set to 'y'.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. Avaya Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

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

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

   Near-end Node Name: clan                  Far-end Node Name: ses-he
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
       Far-end Domain: east.devcon.com

                                             Bypass If IP Threshold Exceeded? n

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

Figure 11: Signaling Group for Outgoing Calls to PSTN

The following signaling group is used for incoming calls from the PSTN. A different signaling group is required because Level 3 specifies a different domain in the FROM header of the SIP INVITE message than what was configured in the far-end domain name field of the signaling group shown in **Figure 11**. The **Far-end Domain** field was left blank, which would match any domain sent by Level 3. In the test configuration, the IP address of the Broadsoft Application Server (AS) was sent as the domain for calls originated from the PSTN. Configuring that IP address in the **Far-end Domain** field is also supported. Follow the instructions described for the signaling group configured above for the 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: clan                   Far-end Node Name: ses-he
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1
       Far-end Domain:

                                              Bypass If IP Threshold Exceeded? n

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

Figure 12: Signaling Group for Incoming Calls from PSTN

Configure the **Trunk Group** form as shown in **Figure 13**. This trunk group is used for outgoing calls to the PSTN. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. For a call between the PSTN and a SIP endpoint, two trunk members are used for the duration of the call. For a call between the PSTN and a non-SIP endpoint, one trunk member is used for the duration of the call. Configure the other fields in bold and accept the default values for the remaining fields.

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

Group Number: 2                    Group Type: sip          CDR Reports: y
  Group Name: Calls to PSTN               COR: 1        TN: 1        TAC: 102
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

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

Figure 13: Trunk Group for Outgoing Calls to PSTN – Page 1

On Page 3 of the trunk group form, set the **Numbering Format** field to *public*. This field specifies the format of the calling party number sent to the far-end.

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

                     Numbering Format: public
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n

 Show ANSWERED BY on Display? y
```

Figure 14: Trunk Group for Outgoing Calls to PSTN – Page 2

Repeat the trunk group configuration in **Figure 13** and **Figure 14** for the trunk group used for incoming calls from the PSTN. The only difference would be to specify the signaling group configured in **Figure 12** for this trunk group. All other fields may be entered as shown.

Note: To call an endpoint on the Avaya SIP-based network from the PSTN, a 10-digit DID number is dialed. This 10-digit dialed number is received by Avaya Communication Manager and converted to the appropriate 5-digit extension in the **Incoming Call Handling Table** (not shown) for trunk group '1'.

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

Group Number: 1                  Group Type: sip           CDR Reports: y
  Group Name: Calls from PSTN            COR: 1        TN: 1        TAC: 101
   Direction: two-way       Outgoing Display? y
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

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

Figure 15: Trunk Group for Incoming Calls from PSTN

Configure the **Public/Unknown Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '6' and whose calls are routed over SIP trunk group '2' have the number sent to the far-end for display purposes. In the example shown in **Figure 16**, a CPN prefix is added to the 5-digit extension so that a 10-digit calling party number (e.g., extension 61003 is converted to 8478761003) is sent to the far-end.

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

 5  6      2         84787          10
```

Figure 16: Public Unknown Format Form

The first step in configuring an off-PBX station (OPS) for the Avaya SIP telephones registered with Avaya SIP Enablement Services is to add a station with the appropriate station type as shown in **Figure 17**. A descriptive name may also be provided. The Class of Restriction (COR) and Class of Service (COS) assigned to the station should be configured with the appropriate call restrictions. Repeat this step for each SIP endpoint at the enterprise site.

```
add station 60004                                           Page   1 of   6
                                     STATION

Extension: 60004                         Lock Messages? n             BCC: 0
     Type: 9630                          Security Code:                TN: 1
     Port: IP                          Coverage Path 1:               COR: 1
     Name: SIP-60004                   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: 60004
            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 17: SIP Station – Page 1

On Page 2 of the station form, verify that the **Per Station CPN – Send Calling Number** field is set to 'y' or blank to allow calling party number information to be sent to the far-end when placing outgoing calls from this station. The default value for this field is blank.

```
add station 60004                                           Page   2 of   6
                                     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? y
  Active Station Ringing: single
                                                       EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
    MWI Served User Type: sip-adjunct          Display Client Redirection? n
                                              Select Last Used Appearance? n
                                                Coverage After Forwarding? s
                                                  Multimedia Early Answer? n
                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 60004         Always Use? n IP Audio Hairpinning? n
```

Figure 18: SIP Station – Page 2

On Page 4 of the station form, configure the appropriate number of call appearances for the SIP telephone. For example, the Avaya 9630 SIP Telephone was configured to support three call appearances as shown in **Figure 19**.

```
add station 60004                                            Page   4 of   6
                                     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:

    voice-mail Number:
```

Figure 19: SIP Station – Page 4

The second step of configuring an off-PBX station is to configure the **Stations with Off-PBX Telephone Integration** form so that calls destined for a SIP telephone at the enterprise site are routed to Avaya SIP Enablement Services, which will then terminate the call to the SIP telephone. On this form, specify the extension of the SIP endpoint and set the **Application** field to *OPS*. The **Phone Number** field is set 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 stations on Avaya Communication Manager. However, this is not a requirement. Finally, the **Trunk Selection** field is set to '2', the SIP trunk group number. This field specifies the trunk group used to route the outgoing call. Another option for routing a call over a SIP trunk group is to use Auto Alternate Routing (AAR) or Auto Route Selection (ARS) routing instead. In this case, the **Trunk Selection** field would be set to *aar* or *ars*. Configuration of other AAR or ARS forms would also be required. Refer to [1] for information on routing calls using AAR or ARS. Repeat this step for each SIP endpoint at the enterprise site.

```
change off-pbx-telephone station-mapping 60004                 Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 60004              OPS          -     60004              2          1
                                 -
```

Figure 20: 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 '3', 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 60004                  Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged        Location
 Extension     Limit      Mode         Allowed     Calls
 60004          3       both           all         both
```

Figure 21: Stations with Off-PBX Telephone Integration – Page 2

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. To access the Administration web interface, enter *http://<ip-addr>/admin* as the URL in an Internet browser, where *<ip-addr>* is the IP address of Avaya SIP Enablement Services. Log in with the appropriate credentials and then select the *Launch Administration Web Interface* link from the **Interface** screen. The main screen shown in **Figure 22** is displayed.

Figure 22: Main Screen

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 assigned during the SES installation process. In the **System Properties** screen, enter the domain name assigned to the Avaya SIP-based network and the SIP License Host. For the **SIP License Host** field, enter the fully qualified domain name or the IP address of the SES server that is running the WebLM application and has the associated license file installed. This entry should always correspond to the localhost unless the WebLM server is not co-resident with this server. After configuring the **System Properties** screen, click the **Update** button.

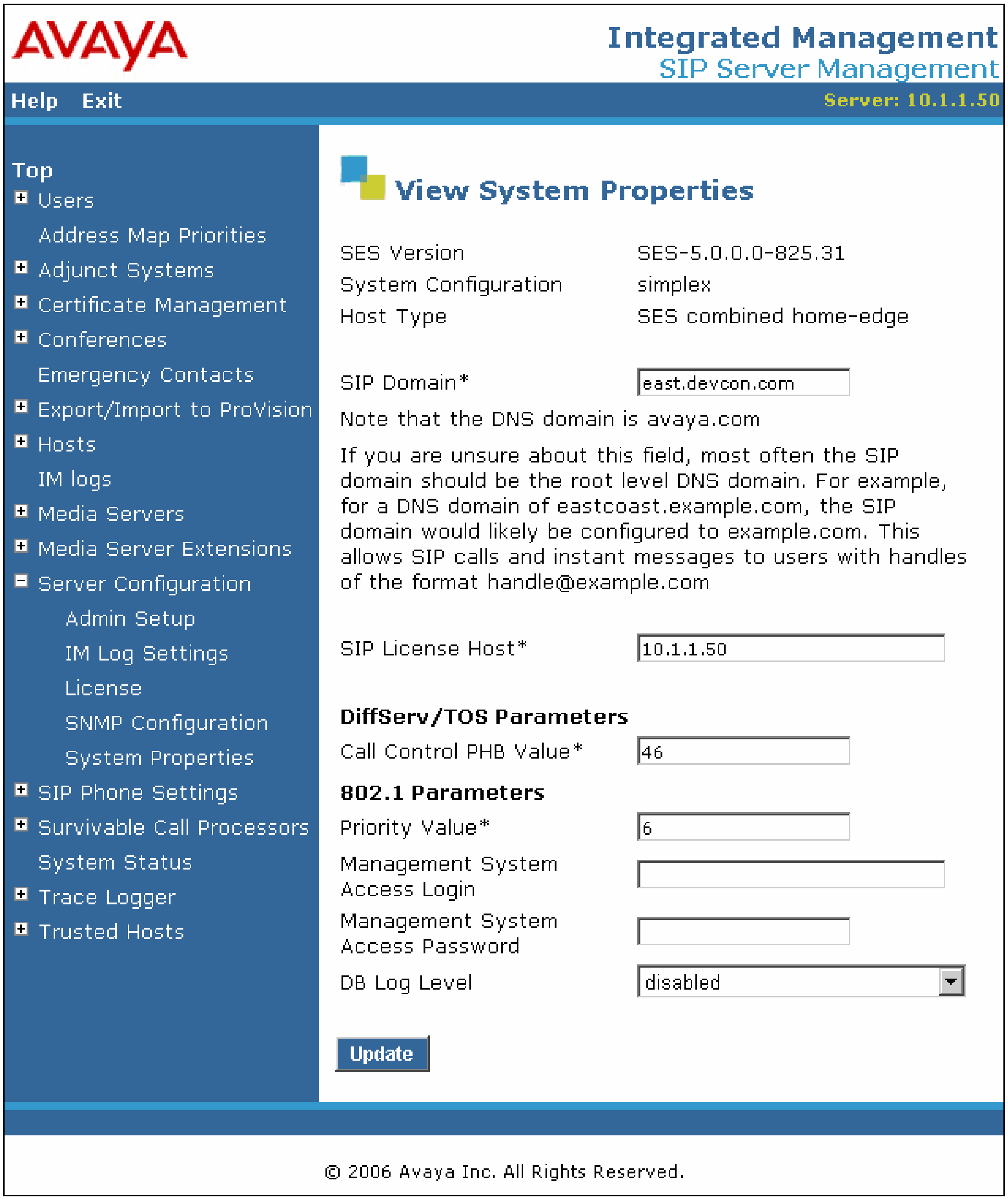

Figure 23: System Properties

After setting up the domain in the **System Properties** screen, create a host entry for Avaya SIP Enablement Services. The following example shows the **Edit Host** screen since the host had already been configured. Enter the IP address of Avaya SIP Enablement Services in the **Host IP Address** field. The **Profile Service Password** was specified during the system installation. Next, configure the **Host Type** field. In this example, the host server was configured as an *SES combined home/edge*. The default values for the other fields may be used as shown in **Figure 24**. Click the **Update** button.

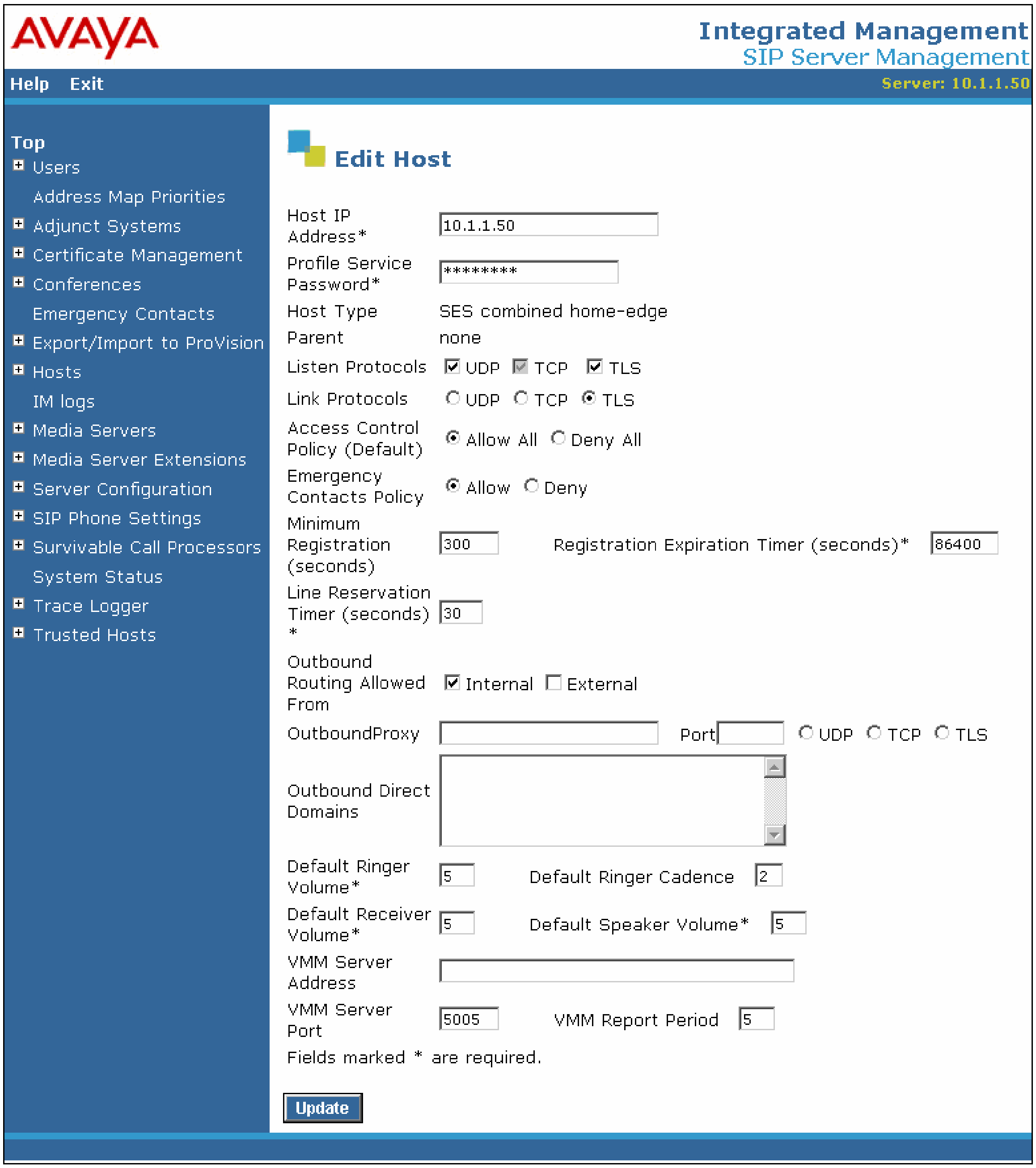

Figure 24: Host

Under the **Media Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server in the enterprise site since a SIP trunk is required between Avaya Communication Manager and Avaya SIP Enablement Services. In the **Add Media Server** screen, enter the following information:

- A descriptive name in the **Media Server Interface** field (e.g., S8710-CLAN).
- Select the home server in the **Host** field.
- Select *TLS* (Transport Link Security) for the **Link Type**. TLS provides encryption at the transport layer.
- Enter the IP address of the C-LAN board in the Avaya G650 Media gateway in the **SIP Trunk IP Address** field.

After completing the **Add Media Server** screen, click the **Add** button. Refer to [3] for additional information on configuring the remaining fields.

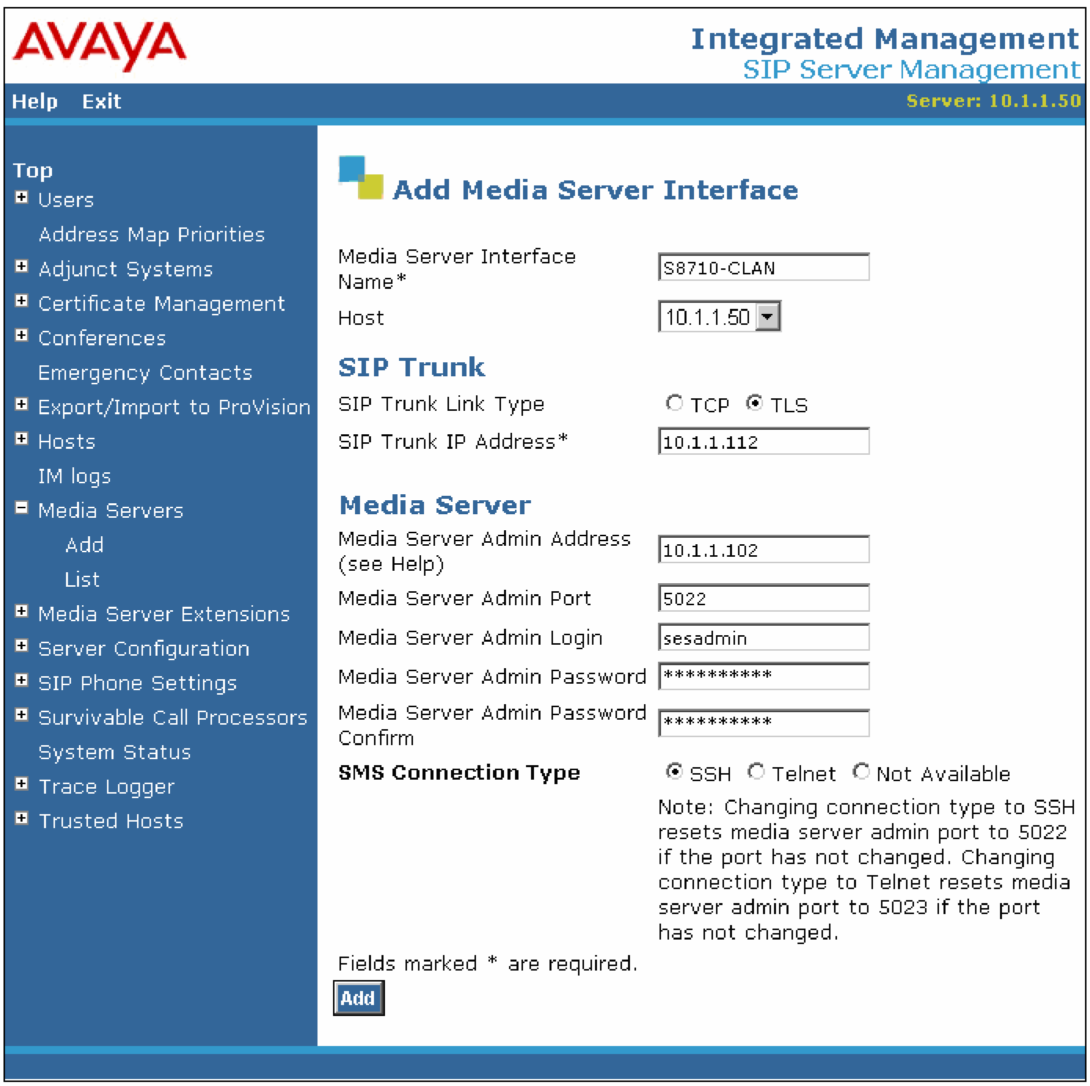

Figure 25: Add Media Server

Incoming calls originated from the PSTN and arriving at Avaya SIP Enablement Services are routed to Avaya Communication Manager for termination services. Calls to be routed to Avaya Communication Manager are specified in a **Media Server Address Map**. The Uniform Resource Identifier (URI) of an incoming INVITE message is compared to the pattern configured in the **Media Server Address Map,** and if there is a match, the call is routed to Avaya Communication Manager. The URI usually takes the form of `sip:user@domain`, where `domain` can be a domain name or an IP address. In this example, `user` is actually the telephone number of the phone. An example of a URI would be `sip:8478792000@20.1.1.10`. Only incoming calls from the PSTN require a media server address map. By default, all calls originated from an Avaya SIP telephone are routed through Avaya Communication Manager for origination services because the Avaya SIP telephones are assigned a media server extension.

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 to display the **List Media Server Address Map** screen and click on the **Add Map In New Group** link. The screen shown in **Figure 26** is displayed. Provide a descriptive name in the **Name** field and enter the regular expression to be used for the pattern matching in the **Pattern** field. In this configuration, the pattern specification matches a URI that begins with `sip:847` followed by seven digits. Note that DID numbers beginning with 847879 were assigned to endpoints at the enterprise site. See Appendix B for a more detailed description of the syntax for address map patterns. Click the **Add** button.

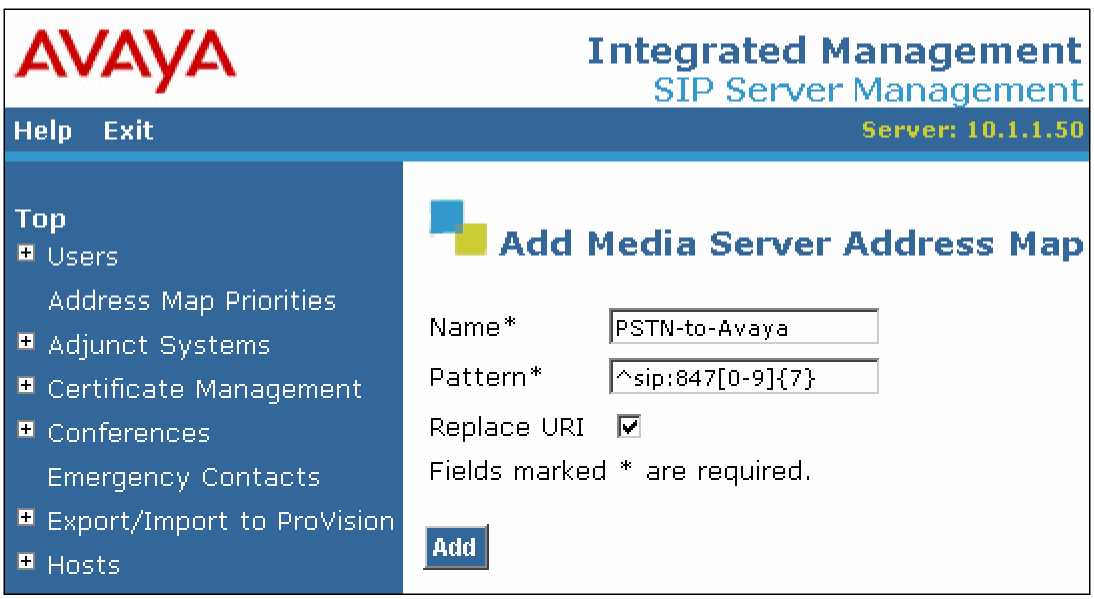

Figure 26: Media Server Address Map

After the **Media Server Address Map** is added, the first **Media Server Contact** is created automatically. For the **Media Server Address Map** added in **Figure 26**, the following contact was created:

```
sip:$(user)@10.1.1.112:5061;transport=tls
```

The contact specifies the IP address of the C-LAN board in the Avaya G650 Media Gateway and the transport protocol used to send SIP signaling messages. The user in the original request URI is substituted for `$(user)`.

After configuring the media server address map, the **List Media Server Address Map** screen appears as shown in **Figure 27**.

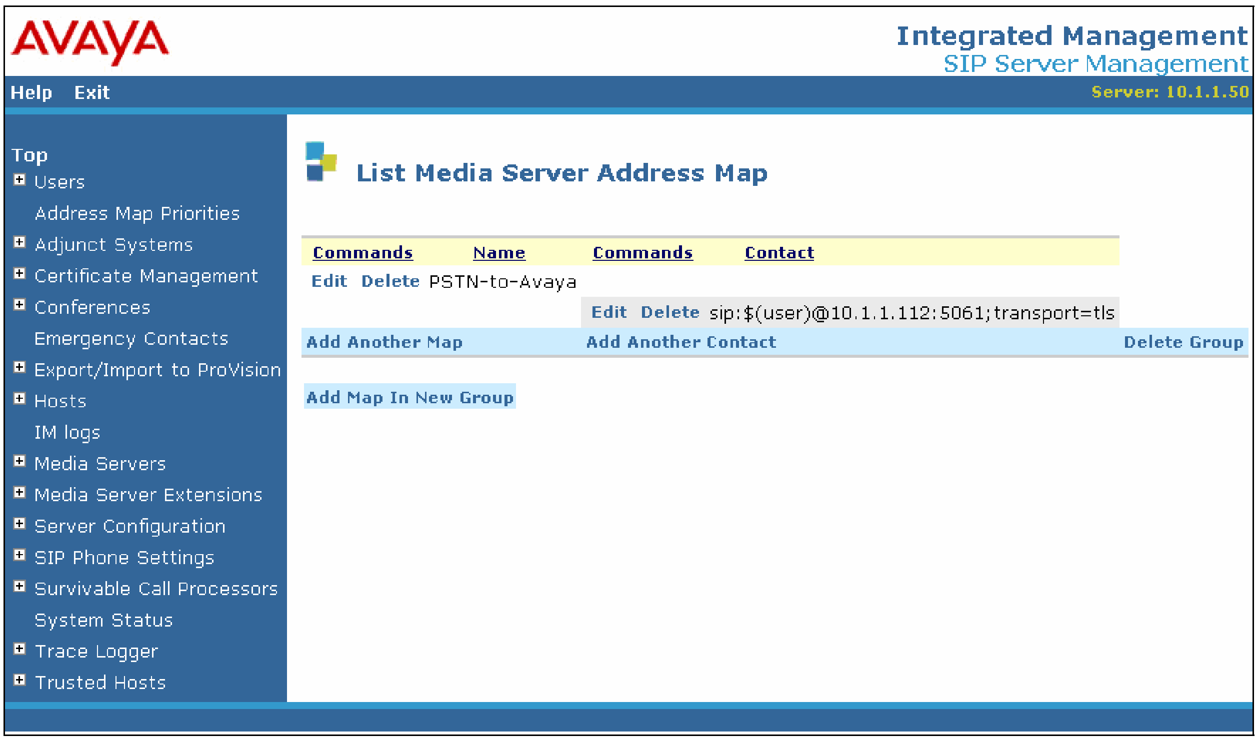

Figure 27: List Media Server Address Map

All calls originated by users at the enterprise site and destined for the PSTN are routed from Avaya SIP Enablement Services to the Level 3 VoIP Network using host address maps. In this configuration, host address maps for the following call types are created. These call types include: calls to area code 732, directory assistance calls, international calls, and toll-free calls.

As an example, the host address map for calls to area code 732 is shown in **Figure 28**. To access the **Add Host Address Map** screen, select the **Hosts** link in the left pane of the Administration web interface and then click on the **Map** link associated with the appropriate host (e.g., 10.1.1.50). The **List Host Address Map** screen is displayed. From this screen, click the **Add Map In New Group** link to display the screen shown in **Figure 28**. Configure a descriptive name for the map and specify an appropriate pattern for the call type. In this example, the pattern is used to route calls to area code 732. By default, the **Replace URI** checkbox is selected. Click the **Add** button.

Figure 28: Add Host Map Entry

From the **List Host Address Map**, click on the **Add Another Contact** link associated with the address map added in **Figure 28**. In this screen, the **Contact** field specifies the destination for the call and it is configured as:

```
sip:$(user)@20.1.1.10:5060;transport=udp
```

The contact specifies the IP address of the Acme Packet Session Border Controller in the Level 3 VoIP Network and the transport protocol used to send SIP signaling messages. The transport protocol must be coordinated with Level 3. The user in the original request URI is substituted for `$(user)`. Click the **Add** button when completed.

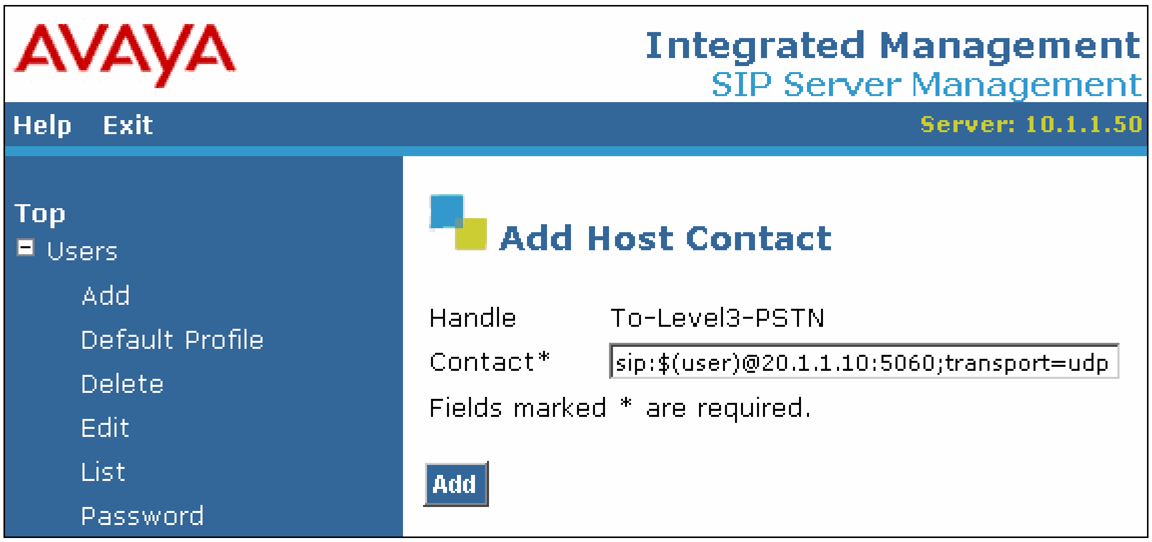

Figure 29: Add Host Contact

After configuring the host address maps, the **List Host Address Map** screen appears as shown in **Figure 30**.

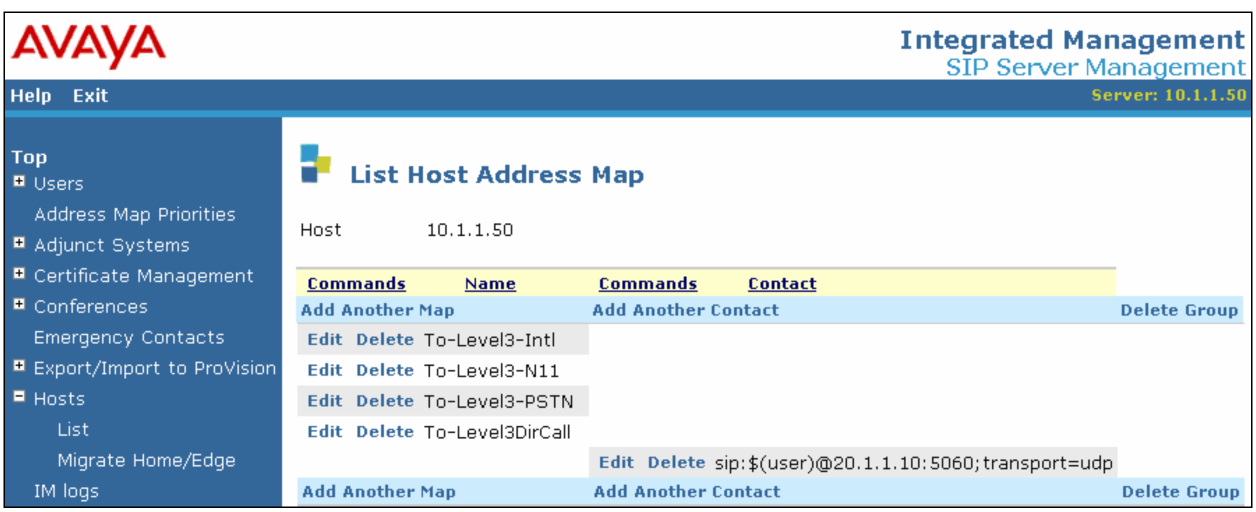

Figure 30: List Host Address Map

Add a user for each Avaya SIP telephone registering with Avaya SIP Enablement Services. 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. In the **Host** field, select the Avaya SIP Enablement Services server hosting the domain (*east.devcon.com*) 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. The **Add Media Server Extension** screen shown in **Figure 32** will be displayed after adding this user profile by clicking on the **Add** button.

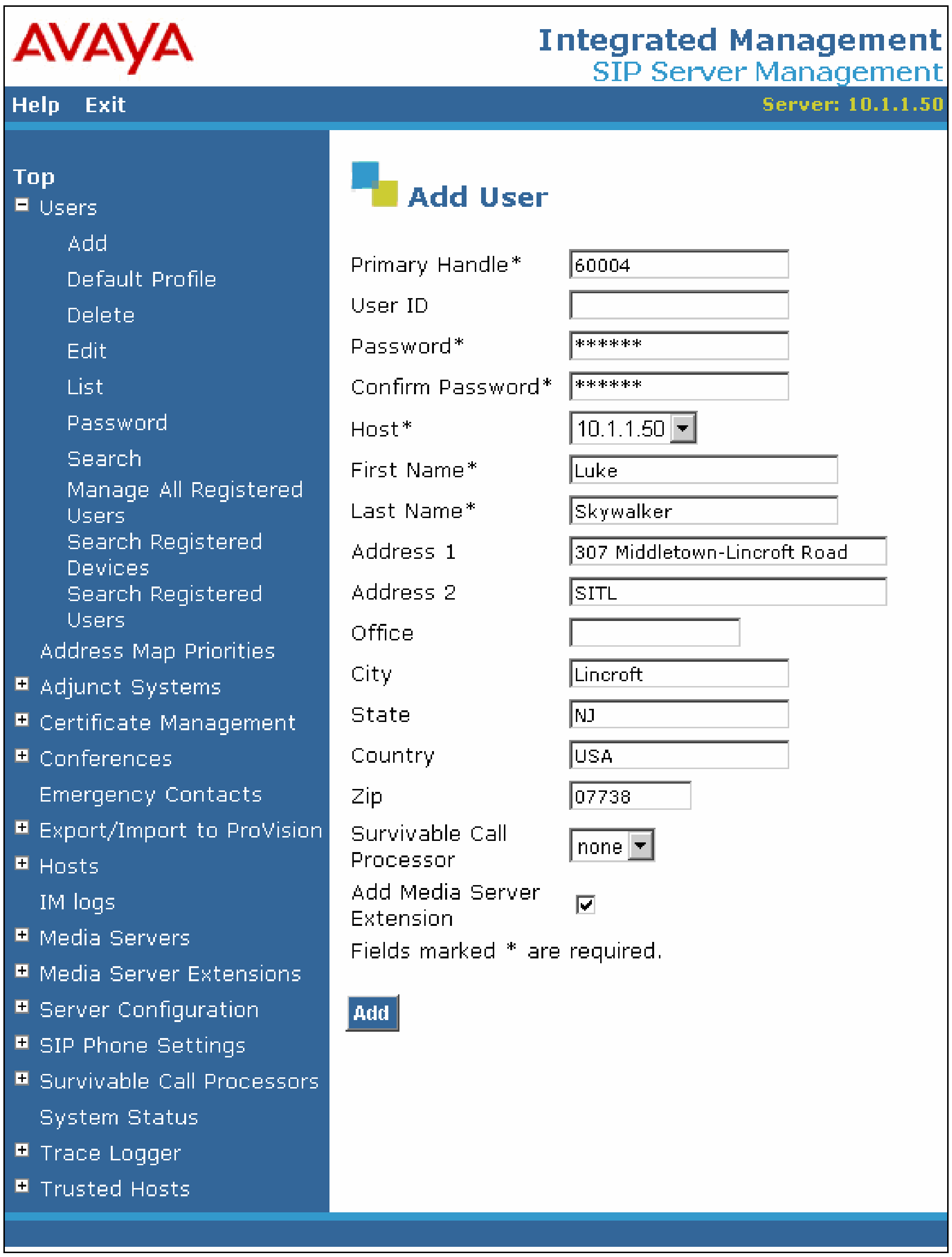

Figure 31: Add User

In the **Add Media Server Extension** screen, enter the **Extension** configured on the media server, shown in **Figure 17**, for the previously added user. Usually, the media server extension and the user extension are the same (recommended). Select the **Media Server** assigned to this extension. Click the **Add** button.

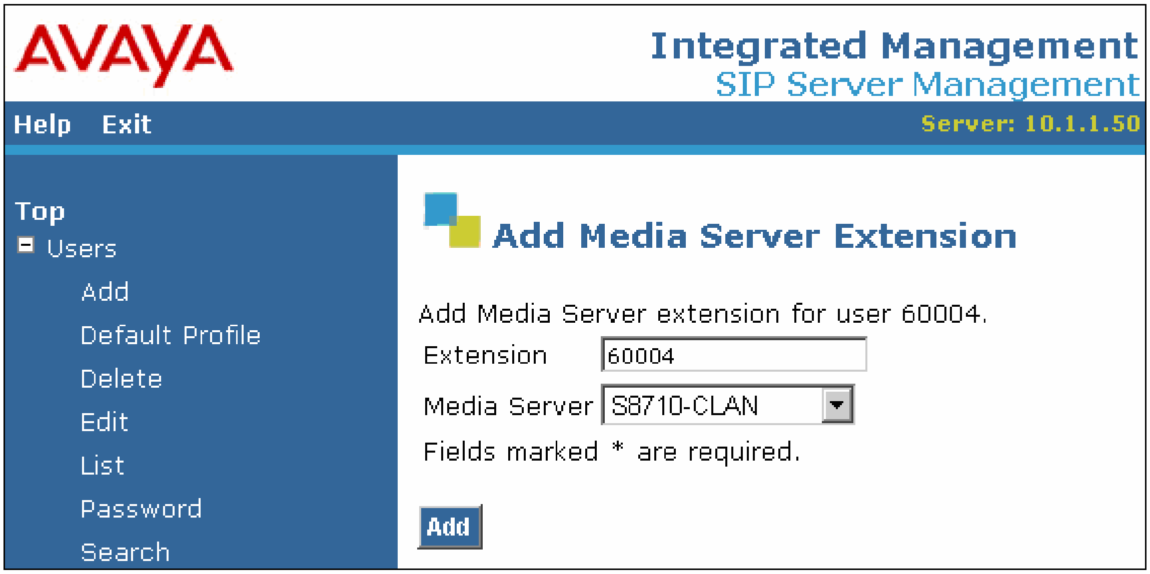

Figure 32: Add Media Server Extension

The last step is to configure the Acme Packet Session Border Controller as a trusted host on Avaya SIP Enablement Services. As a trusted host, Avaya SIP Enablement Services will not issue SIP authentication challenges for incoming requests from the Acme Packet Session Border Controller. Specify the IP address of the Acme Packet SBC in the **IP Address** field and set the Host field to the IP address of Avaya SIP Enablement Services. A descriptive comment can be provided in the **Comment** field.

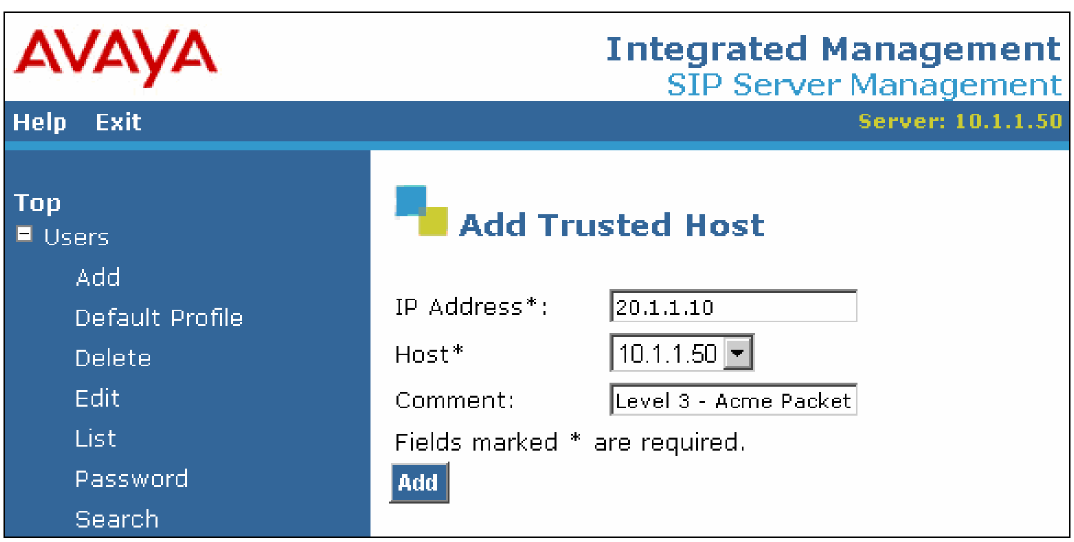

Figure 33: Add a Trusted Host

5. Level 3 VoIP Network Configuration

To use the *Level 3 Enterprise IP Trunking* service, a customer must order the service from Level 3 Communications using their sales processes. The process can be started by contacting Level 3 via their corporate website at http://www.level3.com or by contacting a Level 3 sales representative.

The following table contains the configuration information, coordinated with Level 3, which was used during the interoperability compliance testing to verify the *Level 3 Enterprise IP Trunking* service.

Feature	Test Configuration
Specify Codec(s) Required: ▪ G.711mu-law ▪ G.711a-law ▪ G.729 ▪ RFC2833 DTMF (required)	The network configuration described in these Application Notes was tested with all the codecs (payload types) listed in the left column, except for G.711 a-law. Note that G.729B is not supported by Level 3 for IP-to-PSTN calls. **Note:** RFC2833 is required for shuffling SIP calls.
Define Dial Plan	10-digit dialing, directory assistance calls, toll-free calls, and International (011 Country Code) calls were supported by the test configuration.
Listed Directory Numbers provided by Level 3	Listed directory numbers should be assigned to the endpoints at the enterprise site. This allows calls to be delivered from the PSTN. In this configuration, listed directory numbers beginning with area code 847 were assigned to the SIP, H.323, digital, and analog endpoints in the enterprise network.
Level 3 provides Proxy IP Address	The IP address of the Acme Packet Session Border Controller in the Level 3 VoIP network was 20.1.1.10 and used to configure the host address maps in Avaya SIP Enablement Services.
Customer provides IP Address of Avaya SIP Enablement Services	The IP address of Avaya SIP Enablement Services in the enterprise network was 10.1.1.50. Level 3 used this IP address for routing calls destined to the listed directory numbers assigned to the enterprise site.
SIP Transport Protocol and Port	SIP signaling was transported between Avaya SIP Enablement Services and Level 3 using UDP and port 5060.

6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the Level 3 VoIP network and an Avaya SIP-based network. This section covers the general test approach and the test results.

6.1. General Test Approach

An enterprise site containing an Avaya SIP-based network was interconnected to the Level 3 VoIP network using SIP trunking. The SIP trunk was established between Avaya SIP Enablement Services and an Acme Packet Session Border Controller. This allowed the enterprise site to access the PSTN through the Level 3 VoIP network. The following features and functionality were covered during the SIP trunking interoperability compliance test:

- Incoming calls to the Avaya IP network from the PSTN routed through the Level 3 VoIP network.
- Outgoing calls from the Avaya IP network to the PSTN routed through the Level 3 VoIP network.
- Calls originated and terminated on SIP, H.323, digital and analog endpoints in the Avaya enterprise network.
- Various call types including: local, long distance, international, toll-free, and directory assistance calls.
- Voice calls using G.711 and G.729 codecs, including codec negotiation. Note that G.729B is not supported by Level 3 for IP-to-PSTN calls.
- DTMF transmission using RFC 2833.
- T.38 Fax support.
- Direct IP-to-IP media (also known as “Shuffling” which allows IP endpoints to send audio (RTP) packets directly to each other without using media resources on the Avaya Media Gateway).
- Telephony features including call transfers, conferencing, call forwarding, call hold, and EC500. These features were initiated for PSTN calls.

6.2. Test Results

All test cases passed. Various call types were successfully established between the Avaya IP network and the PSTN. Calls were established using SIP trunking over the Internet to the Level 3 VoIP network.

7. Verification Steps

This section provides verification steps that may be performed in the field to verify that incoming and outgoing PSTN calls can be established between the Avaya IP network and the Level 3 VoIP network.

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call can remain active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
2. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.
3. Verify that the user on the PSTN can terminate an active call by hanging up.
4. Verify that an endpoint at the enterprise site can terminate an active call by hanging up.
5. If Shuffling is enabled, verify that a call originated or terminated on an Avaya IP telephone is shuffled. To determine if the call is shuffled, identify the trunk member active on the call by running the **status trunk <*group*>** command on the SAT of Avaya Communication Manager. Next, run the **status trunk group/member** command and check the **Audio Connection** field. If the call is shuffled, the field should be set to *ip-direct*; otherwise, the field would be set to *ip-tdm*.

8. Support

For technical support on Level 3 Enterprise IP Trunking service, contact the Level 3 Technical Support Center at 1-877-4Level3 (1-877-453-8353).

9. Conclusion

These Application Notes describe the configuration steps required to connect an enterprise site consisting of an Avaya SIP-based Network to the Level 3 VoIP Network. This allows enterprise customers to reduce long distance and interconnection costs by accessing the PSTN through the Level 3 VoIP Network. Enterprise customers subscribing to the Level 3 Enterprise IP Trunking service, can receive and place local, long distance, international, directory assistance, and toll-free calls.

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*, January 2008, Issue 4, Document Number 03-300509.
[2] *SIP Support in Avaya Communication Manager Running on the Avaya S8xxx Servers*, January 2008, Issue 8, Document Number 555-245-206.
[3] *Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services*, January 2008, Issue 5.0, Document Number 03-600768.

Additional information about the Level 3 Enterprise IP Trunking service is available at http://www.level3.com.

APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by the Level 3 VoIP Network 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 Level 3 VoIP Network:

```
Session Initiation Protocol
    Request-Line: INVITE sip:8478789505@20.1.1.10:5060 SIP/2.0
        Method: INVITE
        [Resent Packet: False]
    Message Header
        Via: SIP/2.0/UDP 20.1.1.10:5060;branch=z9hG4bKj680br2090d1mc08d5s1.1
            Transport: UDP
            Sent-by Address: 20.1.1.10
            Sent-by port: 5060
            Branch: z9hG4bKj680br2090d1mc08d5s1.1
        From: <sip:7328821637@172.30.35.193;user=phone>;tag=2137652629-1220908198755-
            SIP from address: sip:7328821637@172.30.35.193
            SIP tag: 2137652629-1220908198755-
        To: "8478789505 8478789505"<sip:8478789505@172.30.35.130:5060>
            SIP Display info: "8478789505 8478789505"
            SIP to address: sip:8478789505@172.30.35.130:5060
        Call-ID: BW2109587550809081473440040@172.30.35.193
        CSeq: 568743346 INVITE
            Sequence Number: 568743346
            Method: INVITE
        Contact: <sip:20.1.1.10:5060;transport=udp>
            Contact Binding: <sip:20.1.1.10:5060;transport=udp>
                URI: <sip:20.1.1.10:5060;transport=udp>
                    SIP contact address: sip:20.1.1.10:5060
        Supported: 100rel
        Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
        Accept: multipart/mixed,application/media_control+xml,application/sdp
        Max-Forwards: 9
        Content-Type: application/sdp
        Content-Disposition: session;handling=optional
        Content-Length: 285
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): BroadWorks 19018 1 IN IP4 20.1.1.10
                Owner Username: BroadWorks
                Session ID: 19018
                Session Version: 1
                Owner Network Type: IN
                Owner Address Type: IP4
                Owner Address: 20.1.1.10
            Session Name (s): -
            Connection Information (c): IN IP4 20.1.1.10
                Connection Network Type: IN
                Connection Address Type: IP4
                Connection Address: 20.1.1.10
            Time Description, active time (t): 0 0
                Session Start Time: 0
                Session Stop Time: 0
            Media Description, name and address (m): audio 11362 RTP/AVP 0 8 18 101
                Media Type: audio
                Media Port: 11362
```

```
        Media Proto: RTP/AVP
        Media Format: ITU-T G.711 PCMU
        Media Format: ITU-T G.711 PCMA
        Media Format: ITU-T G.729
        Media Format: 101
    Media Attribute (a): rtpmap:0 PCMU/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 0
        MIME Type: PCMU
    Media Attribute (a): rtpmap:8 PCMA/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 8
        MIME Type: PCMA
    Media Attribute (a): rtpmap:18 G729/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 18
        MIME Type: G729
    Media Attribute (a): fmtp:18 annexb=no
        Media Attribute Fieldname: fmtp
        Media Format: 18 [G729]
        Media format specific parameters: annexb=no
    Media Attribute (a): rtpmap:101 telephone-event/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 101
        MIME Type: telephone-event
    Media Attribute (a): fmtp:101 0-15
        Media Attribute Fieldname: fmtp
        Media Format: 101 [telephone-event]
        Media format specific parameters: 0-15
    Media Attribute (a): sendrecv
    Media Attribute (a): maxptime:20
        Media Attribute Fieldname: maxptime
        Media Attribute Value: 20
```

Sample SIP INVITE Message from Avaya SIP Enablement Services to Level 3:

```
Session Initiation Protocol
    Request-Line: INVITE sip:17328821637@20.1.1.10:5060;lr SIP/2.0
        Method: INVITE
        [Resent Packet: False]
    Message Header
        Accept-Language: en
        Call-ID: 0f02819b78cdd131e748bd719400
        CSeq: 1 INVITE
            Sequence Number: 1
            Method: INVITE
        From: "H323-61003"
<sip:8478761003@east.devcon.com:5061>;tag=0f02819b78cdd130e748bd719400
            SIP Display info: "H323-61003"
            SIP from address: sip:8478761003@east.devcon.com:5061
            SIP tag: 0f02819b78cdd130e748bd719400
        Record-Route: <sip:10.1.1.50:5060;lr>,<sip:10.1.1.112:5061;lr;transport=tls>
        To: "17328821637" <sip:17328821637@east.devcon.com>
            SIP Display info: "17328821637"
            SIP to address: sip:17328821637@east.devcon.com
        Via: SIP/2.0/UDP
10.1.1.50:5060;branch=z9hG4bK030303666666030303343e.0,SIP/2.0/TLS
10.1.1.112;psrrposn=2;received=10.1.1.112;branch=z9hG4bK0f02819b78cdd132e748bd719400
            Transport: UDP
            Sent-by Address: 10.1.1.50
            Sent-by port: 5060
            Branch: z9hG4bK030303666666030303343e.0,SIP/2.0/TLS
        Content-Length: 168
        Content-Type: application/sdp
        Contact: "H323-61003" <sip:8478761003@10.1.1.112:5061;transport=tls>
            Contact Binding: "H323-61003"
<sip:8478761003@10.1.1.112:5061;transport=tls>
                URI: "H323-61003" <sip:8478761003@10.1.1.112:5061;transport=tls>
                    SIP Display info: "H323-61003"
                    SIP contact address: sip:8478761003@10.1.1.112:5061
        Max-Forwards: 68
        User-Agent: Avaya CM/R015x.00.0.825.4
        Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS,INFO,PUBLISH
        Supported: 100rel,timer,replaces,join,histinfo
        Alert-Info: <cid:internal@east.devcon.com>;avaya-cm-alert-type=internal
        Min-SE: 1800
        Session-Expires: 1800;refresher=uac
        P-Asserted-Identity: "H323-61003" <sip:8478761003@east.devcon.com:5061>
        History-Info: <sip:17328821637@east.devcon.com>;index=1,"17328821637"
<sip:17328821637@east.devcon.com>;index=1.1
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 1 1 IN IP4 10.1.1.112
                Owner Username: -
                Session ID: 1
                Session Version: 1
                Owner Network Type: IN
                Owner Address Type: IP4
                Owner Address: 10.1.1.112
            Session Name (s): -
            Connection Information (c): IN IP4 10.1.1.116
                Connection Network Type: IN
                Connection Address Type: IP4
                Connection Address: 10.1.1.116
            Bandwidth Information (b): AS:64
```

```
        Bandwidth Modifier: AS [Application Specific (RTP session bandwidth)]
        Bandwidth Value: 64 kb/s
    Time Description, active time (t): 0 0
        Session Start Time: 0
        Session Stop Time: 0
    Media Description, name and address (m): audio 24552 RTP/AVP 0 127
        Media Type: audio
        Media Port: 24552
        Media Proto: RTP/AVP
        Media Format: ITU-T G.711 PCMU
        Media Format: 127
    Media Attribute (a): rtpmap:0 PCMU/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 0
        MIME Type: PCMU
    Media Attribute (a): rtpmap:127 telephone-event/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 127
        MIME Type: telephone-event
```

APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within 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 characters.

The pattern matching string used in Avaya SES may use any of the following metacharacters:

- 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:17325551638@20.1.1.10:5060;transport=udp 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.