Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between Cincinnati Bell Any Distance eVantage with an Avaya IP Telephony Network - Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between Cincinnati Bell Any Distance eVantage and an Avaya IP Telephony Network consisting of Avaya Aura™ SIP Enablement Services and Avaya Aura™ Communication Manager. Avaya IP, digital and analog endpoints were used to originate and terminate calls.

Cincinnati Bell, Inc. is a member of the Avaya DevConnect Service Provider program. 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 Cincinnati Bell Any Distance (CBAD) eVantage solution and an Avaya IP Telephony Network consisting of Avaya Aura™ SIP Enablement Services and Avaya Aura™ Communication Manager. Avaya IP, digital and analog endpoints were used to originate and terminate calls.

SIP (Session Initiation Protocol) is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [4] is the primary specification governing this protocol. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc. Within these Application Notes, SIP is used as the signaling protocol between SIP Enablement Services and the network services offered by Cincinnati Bell Any Distance eVantage solution.

The CBAD eVantage solution is a turn-key business trunking solution for customers. eVantage provides customers with a single IP connection that converges voice and data services to drive optimization, reduce costs, and offer enhanced features not typically available in the traditional PSTN network. Voice services, such as local, long distance, and toll free calling, as well a high speed data and Internet services, are the primary applications of the eVantage solution.

1.1. Interoperability Compliance Testing

A simulated enterprise site consisting of a Communication Manager and SIP Enablement Services solution supporting SIP trunking was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the commercially available SIP trunking solution provided by CBAD eVantage solution. This allowed the enterprise site to use SIP trunking for calls to the 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 Cincinnati Bell.
- Outgoing calls from the enterprise site were completed via CBAD eVantage solution to PSTN destinations.
- Calls using H.323, SIP, digital and analog endpoints supported by the Avaya IP Telephony network.
- Various call types including: local, long distance, international, and toll free calls.
- Calls using the G.729(a) and G.711 μLAW codecs.
- DTMF tone transmission using RFC 2833 with successful call vectoring application.
- Telephone features such as hold, transfer, conference, and call forwarding.
- Avaya one-X® Communicator in either Telecommuter or Softphone mode.

1.2. Support

For technical support on Cincinnati Bell Any Distance eVantage solution, customers can call 1-866-914-9474.

2. Reference Configuration

Figure 1 illustrates an enterprise site with an Avaya SIP-based network, including SIP Enablement Services, a S8500C Server with a G650 Media Gateway[1] running Communication Manager, and Avaya IP, digital, and analog endpoints. The enterprise site is connected to the Cincinnati Bell Any Distance eVantage solution over the Internet and communicates using SIP. The CBAD Network is accessible via a Cisco CUBE supporting a public IP address of 16.96.81.46.

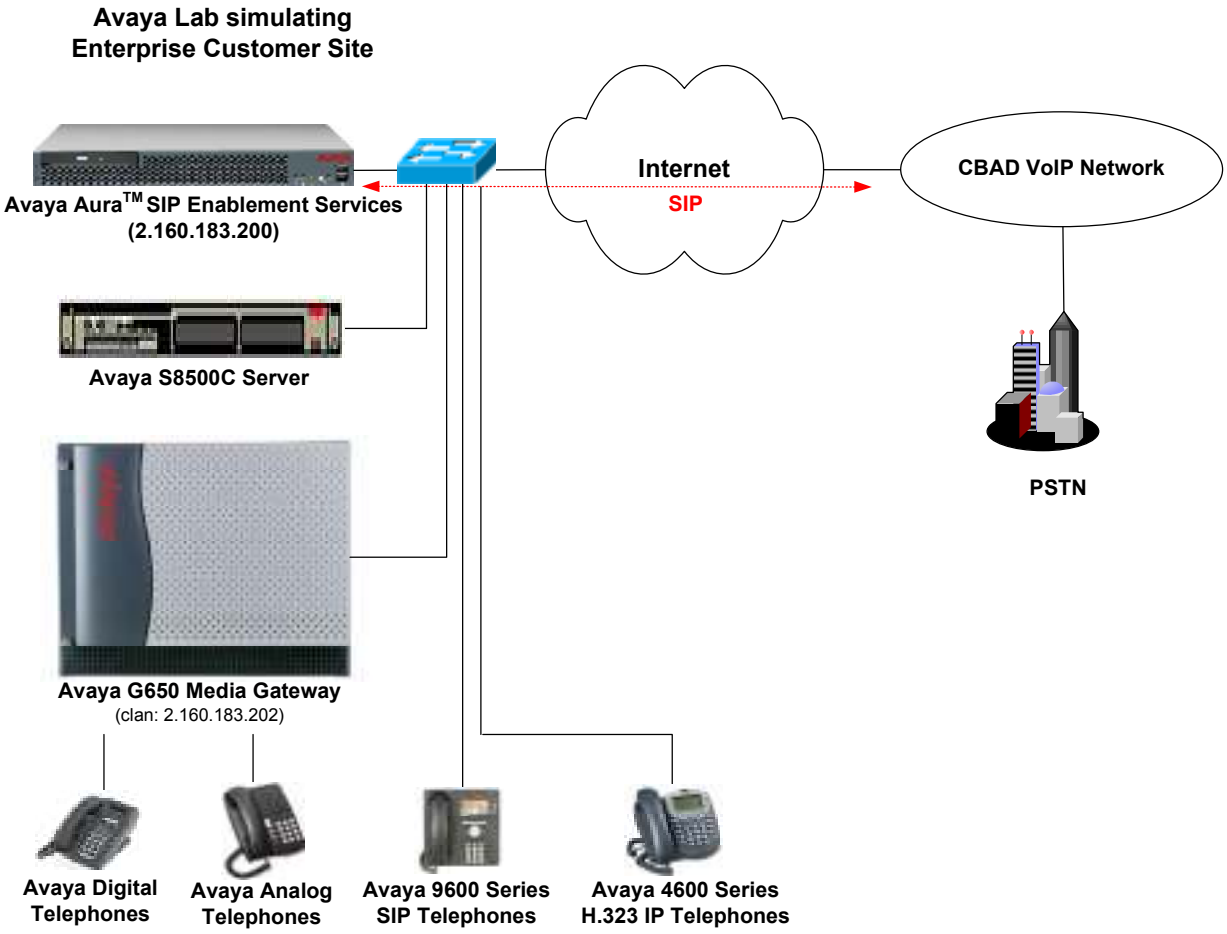

Figure 1: Avaya IP Telephony Network connected to CBAD eVantage Solution

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

2.1. 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 CBAD 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 CBAD eVantage solution so the call is routed through the CBAD VoIP network.
2. Based on the DID number, CBAD routes the call to the enterprise site via SIP trunking. CBAD sends SIP signaling messages to SIP Enablement Services at the enterprise site. See the Appendix A for an example of a SIP INVITE message sent by CBAD.
3. SIP Enablement Services routes the call to the Avaya S8720 Server running Communication Manager over a SIP trunk.
4. Since the call is destined for an Avaya SIP telephone, Communication Manager routes the call back to SIP Enablement Services over a SIP trunk. If the destination of the call was an H.323, digital or analog endpoint, Communication Manager would terminate the call directly to the endpoint and steps 4 and 5 would not be required.
5. SIP Enablement Services terminates the call to the Avaya SIP telephone.

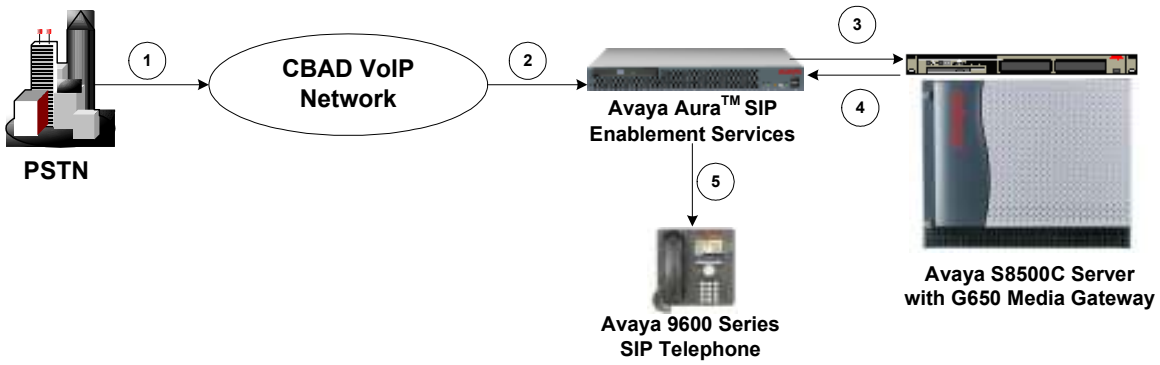

Figure 2: PSTN Call to the Avaya SIP Network

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 SIP Enablement Services. If the originator were an H.323, digital or analog endpoint, the call request would be sent to SIP Enablement Services from the S8720 Servers running Communication Manager.
2. SIP Enablement Services routes the call over the SIP trunk to the Avaya S8720 Servers running Communication Manager for origination services. This allows 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, Communication Manager routes the call back to Avaya SIP Enablement Services over a SIP trunk.
4. SIP Enablement Services routes the call to the CBAD VoIP Network. See Appendix A for an example of a SIP INVITE message sent by the Avaya SIP-based network.
5. CBAD routes the call to the PSTN.

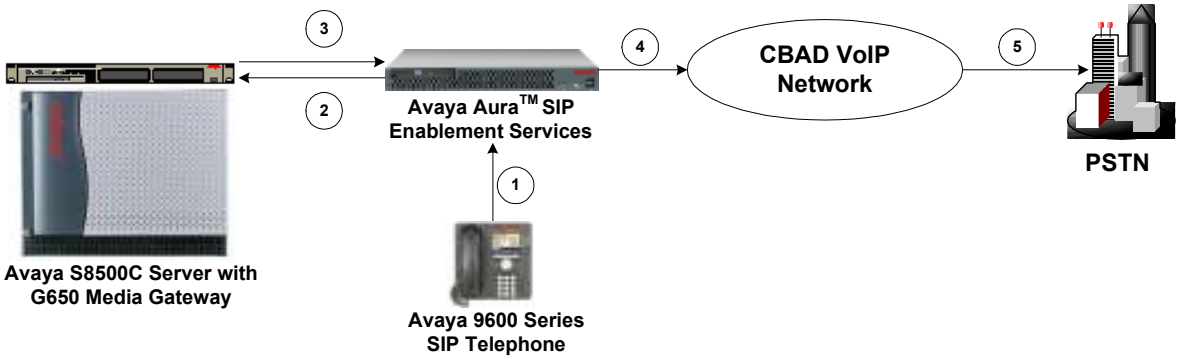

Figure 3: Avaya SIP Call to the PSTN

3. Equipment and Software Validated

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

Hardware Component	Version
Avaya S8500C Server	Avaya Aura Communication Manager 5.2 (R015x.002.0.947.3) with Service Pack 1 (Update 17294)
Avaya G650 Media Gateway ▪ TN799DP C-LAN Board ▪ TN2302AP Media Processor Board	 HW13 FW032 HW21 FW120
Avaya Aura SIP Enablement Services	5.2 (SES05.2-02.0.947.3b) with Service Pack 1(SES-02.0.947.3-SP1)
Avaya 1600 Series IP Telephone	1.1 (H.323)
Avaya 4600 Series IP Telephones	2.9 (H.323); R2.2.2 (SIP)
Avaya 9600 Series IP Telephones	3.0.2 (H.323) 2.4.1.0 (SIP)
Avaya Digital Telephones	--
Avaya Analog Telephones	--
Avaya one-X Communicator	R1.030-SP3-16918
Cisco Cube	12.4 (24)T1

4. Configure Communication Manager

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

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), International calls (011 Country Code), Toll-Free calls, and Operator calls were also supported. 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 **Error! Reference source not found.** for additional information.

Communication Manager configuration was performed using the System Access Terminal (SAT). The IP network parameters of the Avaya S8500C Server 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  11
                            OPTIONAL FEATURES

     G3 Version: V15                              Software Package: Standard
       Location: 1                             RFA System ID (SID): 1
       Platform: 12                            RFA Module ID (MID): 1

                                                          USED
                                Platform Maximum Ports: 44000 226
                                     Maximum Stations: 36000 80
                                Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 10    1
                    Maximum Off-PBX Telephones -   OPS: 300   55
                    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  11
                               OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 100    6
          Maximum Concurrently Registered IP Stations: 18000 1
            Maximum Administered Remote Office Trunks: 0      0
Maximum Concurrently Registered Remote Office Stations: 0      0
             Maximum Concurrently Registered IP eCons: 0      0
  Max Concur Registered Unauthenticated H.323 Stations: 0      0
                 Maximum Video Capable H.323 Stations: 0      0
                  Maximum Video Capable IP Softphones: 0      0
                     Maximum Administered SIP Trunks: 600  130
  Maximum Administered Ad-hoc Video Conferencing Ports: 0      0
   Maximum Number of DS1 Boards with Echo Cancellation: 0      0
                           Maximum TN2501 VAL Boards: 10     0
                    Maximum Media Gateway VAL Sources: 0      0
          Maximum TN2602 Boards with 80 VoIP Channels: 128    0
         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 2.1 did not require trunk-to-trunk transfer to be enabled.

```
change system-parameters features                               Page   1 of  18
                           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 SIP Enablement Services at the enterprise site. The host names will be used throughout the other configuration screens of Communication Manager.

```
change node-names ip                                          Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
clan1               2.160.183.202
medpro1             2.160.183.203
ses                 2.160.183.200
default             0.0.0.0

( 4 of 12   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 SIP Enablement Services. In this configuration, the domain name is *avremote.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 10**. 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 SIP Enablement Services. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the **Far-end Network Region** field of the SIP signaling group as shown in **Figure 10**.

```
change ip-network-region 1                                    Page   1 of  19
                             IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avremote.com
    Name: Main
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: 60001
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.

```
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: G.729A           n          2        20
 3:
 4:
 5:
 6:
 7:
```

Figure 9: IP Codec Set – Page 1

Prior to configuring a SIP trunk group for communication with SIP Enablement Services, a SIP signaling group must be configured. The following signaling group is used for outgoing calls to the PSTN through the CBAD eVantage solution. Configure the Signaling Group form shown in **Figure 10** 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 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 for PSTN calls could have been specified.
- Enter the domain name of SIP Enablement Services in the **Far-end Domain** field. The **Far-end Domain** field was filled with the IP address of the Cisco CUBE SIP proxy/SBC, *16.96.81.46*. The **Far-end Domain** field 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*. Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 11                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 11              Group Type: sip
                         Transport Method: tls


   Near-end Node Name: clan1                 Far-end Node Name: ses
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                       Far-end Network Region: 1
       Far-end Domain: 16.96.81.46

                                            Bypass If IP Threshold Exceeded? n

        DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? Y
Session Establishment Timer(min): 3                    IP Audio Hairpinning? n
        Enable Layer 3 Test? n
H.323 Station Outgoing Direct Media? n           Alternate Route Timer(sec): 6
```

Figure 10: 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 CBAD 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. The **Far-end Domain** field was filled with *as.voip.fuse.net*, which would match the domain sent by CBAD. Follow the instructions described for the signaling group configured above for the other fields.

```
add signaling-group 10                                        Page   1 of   1
                               SIGNALING GROUP

 Group Number: 10             Group Type: sip
                         Transport Method: tls

   Near-end Node Name: clan1                 Far-end Node Name: ses
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                         Far-end Network Region: 1
       Far-end Domain: as.voip.fuse.net

                                              Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? Y
Session Establishment Timer(min): 3                    IP Audio Hairpinning? n
         Enable Layer 3 Test? n
H.323 Station Outgoing Direct Media? n           Alternate Route Timer(sec): 6
```

Figure 11: Signaling Group for Incoming Calls from the PSTN

Configure the Trunk Group form as shown in **Figure 12**. This trunk group is used for outgoing calls to the PSTN. Set the **Group Type** field to *sip*, set the **Service Type** field to *public-ntwrk*, 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 11                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 11                   Group Type: sip           CDR Reports: y
  Group Name: CBTS Outgoing                 COR: 1        TN: 1        TAC: 111
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                     Signaling Group: 11
                                                   Number of Members: 30
```

Figure 12: Trunk Group for Outgoing Calls to PSTN – Page 1

On Page 2 of the trunk group form, set the **Preferred Minimum Session Refresh Interval(sec)** to the agreed upon rate discussed with CBAD. The value of *900* was chosen for testing purposes.

```
add trunk-group 11                                          Page   2 of  21
         Group Type: sip

TRUNK PARAMETERS

     Unciode Name? y

                                        Redirect on OPTIM Failure: 5000

            SCCAN? N                              Digital Loss Group: 18
                     Preferred Minimum Session Refresh Interval(sec): 900
```

Figure 13: Trunk Group for Outgoing Calls to PSTN – Page 2

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 11                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y

                     Numbering Format: public
                                              UUI Treatment: service-provider

                                               Replace Restricted Numbers? y
                                              Replace Unavailable Numbers? y

Show ANSWERED BY on Display? y
```

Figure 14: Trunk Group for Outgoing Calls to PSTN – Page 3

On Page 4 of the trunk group form, set the **Telephone Event Payload Type** to *101* for the proper exchange of RFC 2833 DTMF events between Communication Manager and the CBAD eVantage solution. Set the fields **Send Transferring Party Information** and **Overwrite Calling Identity** to *y* to permit trunk to trunk transfer to work correctly.

```
add trunk-group 11                                          Page   4 of  21
                            PROTOCOL VARIATIONS

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

                   Send Diversion Header? n
                 Support Request History? n
             Telephone Event Payload Type: 101

               Overwrite Calling Identity? y
```

Figure 15: Trunk Group for Outgoing Calls to PSTN – Page 4

Repeat the trunk group configuration in **Figure 12** through **Figure 15** for the trunk group used for incoming calls from the PSTN. The only difference would be to specify the signaling group configured in **Figure 11** 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 Communication Manager and converted to the appropriate 5-digit extension in the **Incoming Call Handling Table** (not shown) for trunk group '10'.

```
add trunk-group 10                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 10                    Group Type: sip           CDR Reports: y
  Group Name: CBTS incoming               COR: 1        TN: 1        TAC: 110
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                      Signaling Group: 10
                                                    Number of Members: 30
```

Figure 16: 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 the local stations 68010 and 68020 who have calls routed over the SIP trunk group *11* will send the number expected by the CBAD eVantage solution for call authentication and for display purposes to the far-end. In the example shown in **Figure 17,** a CPN prefix is added to the 5-digit extension so that a 10-digit calling party number (e.g., extension 68010 is converted to 5135551234) is sent to the far-end.

Note: The 10-digit CPN must be recognized by the CBAD VoIP network or the call will be denied.

```
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: 5
 5  68010         11        5135551234     10        Maximum Entries: 9999
 5  68020         11        5135551235     10
```

Figure 17: Public Unknown Format Form

The first step in configuring an off-PBX station (OPS) for the Avaya SIP telephones registered with SIP Enablement Services is to add a station with the appropriate station type as shown in **Figure 18**. 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 68020                                          Page   1 of   6
                                  STATION

Extension: 68020                      Lock Messages? n               BCC: 0
     Type: 9600SIP                    Security Code:                  TN: 1
     Port: S00009                   Coverage Path 1:                 COR: 1
     Name: Johnny SIP               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: 68020
           Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english              Expansion Module? 0
 Survivable GK Node Name:
          Survivable COR: internal            Media Complex Ext:
   Survivable Trunk Dest? y                        IP SoftPhone? n




                                           Customizable Labels? y
```

Figure 18: 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 68020                                          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:                    Display Client Redirection? n
            AUDIX Name:                     Select Last Used Appearance? n
                                              Coverage After Forwarding? s

                                           Direct IP-IP Audio Connections? y
  Emergency Location Ext: 20003        Always Use? n IP Audio Hairpinning? n
```

Figure 19: 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 20**.

```
add station 68020                                          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:
```

Figure 20: 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 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 SIP Enablement Services also match the extensions of the corresponding stations on Communication Manager. However, this is not a requirement. Finally, the **Trunk Selection** field is set to *1*, 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. If either option is preferred, the **Trunk Selection** field would be set to *aar* or *ars*. Configuration of other AAR or ARS forms would also be required. Refer to **Error! Reference source not found.** 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 68020                 Page   1 of   3
                 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                      Selection   Set
 68020            OPS           -     68020              1          1
```

Figure 21: 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 68020                  Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 68020         OPS      3        both         all         none
```

Figure 22: Stations with Off-PBX Telephone Integration – Page 2

5. Configure SIP Enablement Services

This section covers the administration of SIP Enablement Services (SES). 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 SIP Enablement Services. Log in with the appropriate credentials and then select the *SIP Enablement Services* from the *Administration* pull down menu from the title bar. The main screen shown in **Figure 23** is displayed.

Figure 23: Main Screen

From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. In the **View 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 **View System Properties** screen, click the **Update** button.

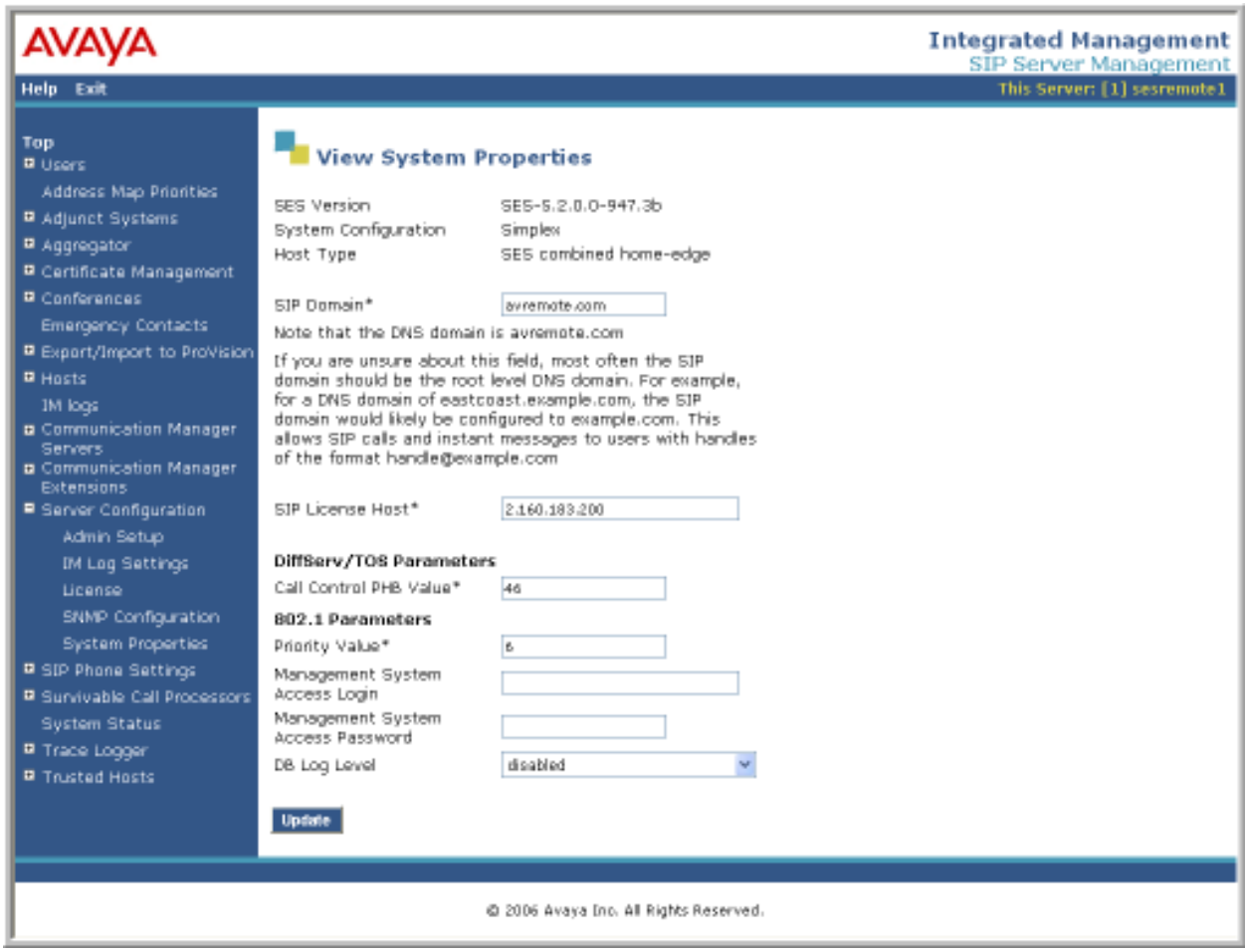

Figure 24: System Properties

After setting up the domain in the **View System Properties** screen, create a host entry for SIP Enablement Services. The following example shows the **Edit Host** screen since the host had already been configured. Enter the IP address of 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 25**. Click the **Update** button.

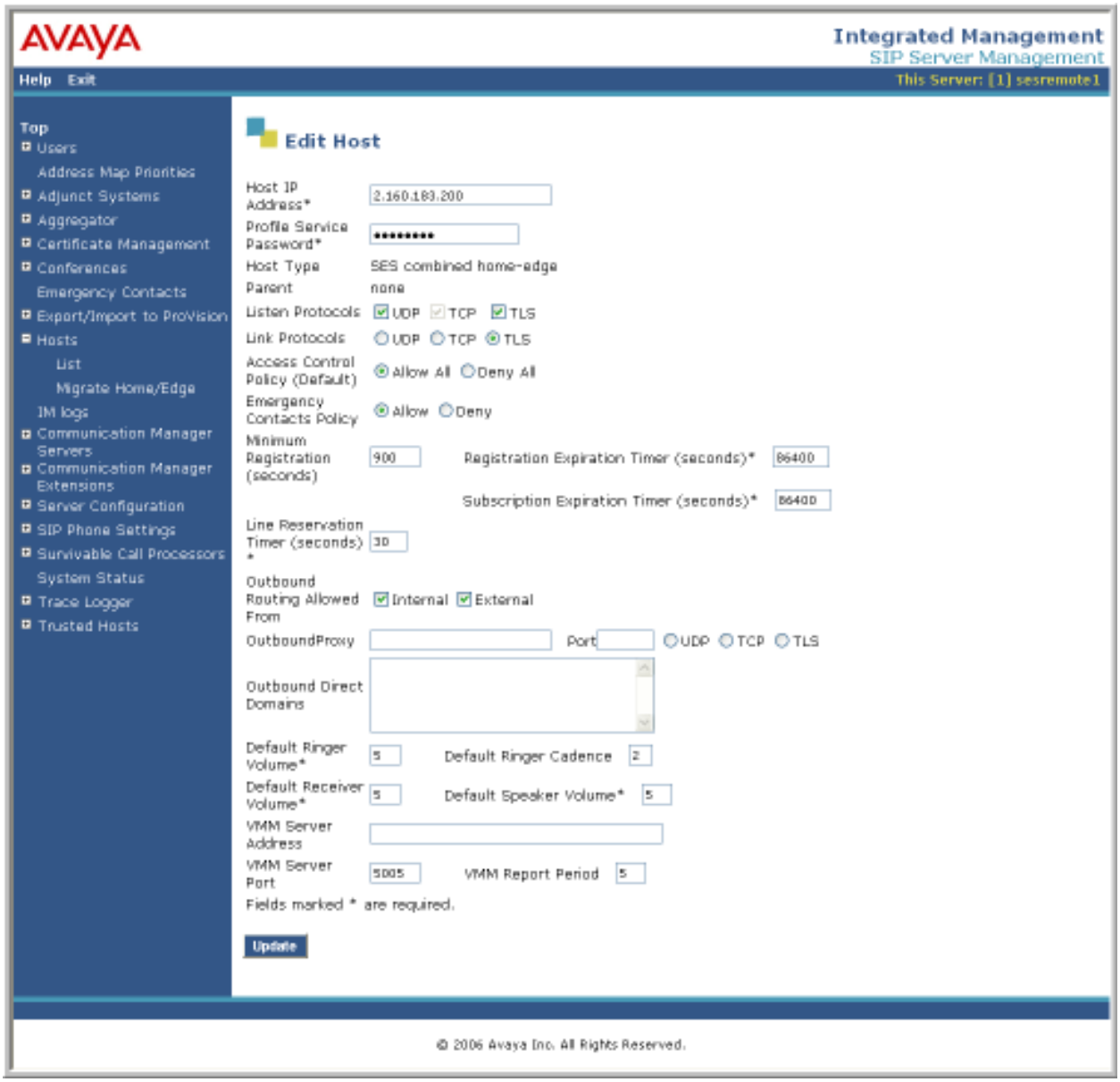

Figure 25: Host

Under the **Communication Manager Servers** option in the Administration web interface, select **Add** to add the Avaya S8500C Server in the enterprise site since a SIP trunk is required between Communication Manager and SIP Enablement Services. 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., CLAN1).
- 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 Communication Manager Server Interface** screen, click the **Add** button. Refer to [3] for additional information on configuring the remaining fields.

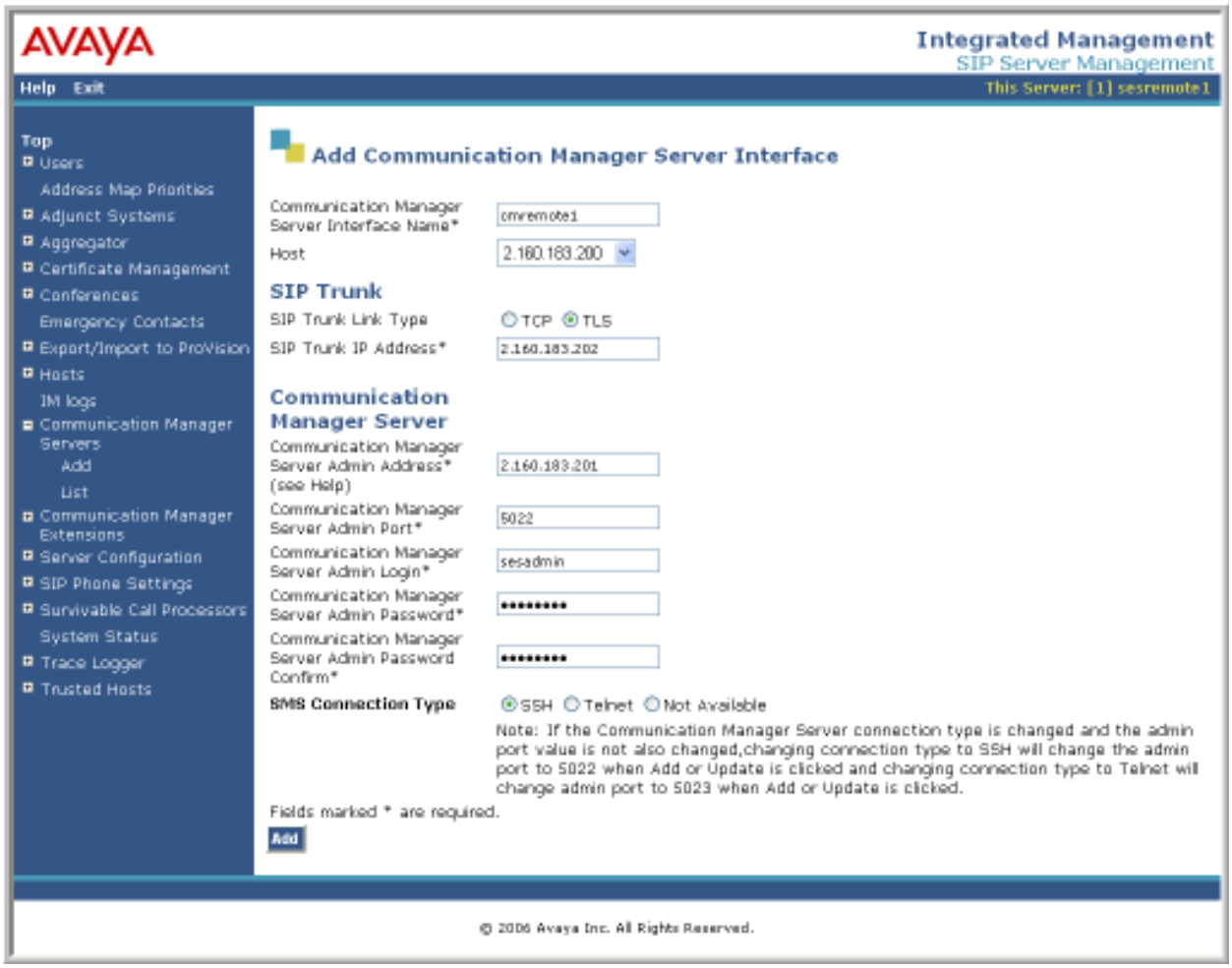

Figure 26: Add Communication Manager Server Interface

Incoming calls originated from the PSTN and arriving at SIP Enablement Services are routed to Communication Manager for termination services. Calls to be routed to Communication Manager are specified in a **Communication Manager Server Address Map**. The Uniform Resource Identifier (URI) of an incoming INVITE message is compared to the pattern configured in the address map, and if there is a match, the call is routed to 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:5135551234@2.160.183.202`. Only incoming calls from the PSTN require a Communication Manager address map. By default, all calls originated from an Avaya SIP telephone are routed through Communication Manager for origination services because the Avaya SIP telephones are assigned a media server extension.

To configure a **Communication Manager Server Address Map**, select **Communication Manager Servers** in the left pane of the Administration web interface. This will display the **List Communication Manager Servers** screen. Click on the **Map** link associated with the appropriate server to display the **List Communication Manager Server Address Map** screen and click on the **Add Map In New Group** link. The screen shown in **Figure 27** 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:513` followed by seven digits. Note that DID numbers beginning with area code 513 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. Repeat this procedure to add an address map for routing incoming toll-free calls, if necessary.

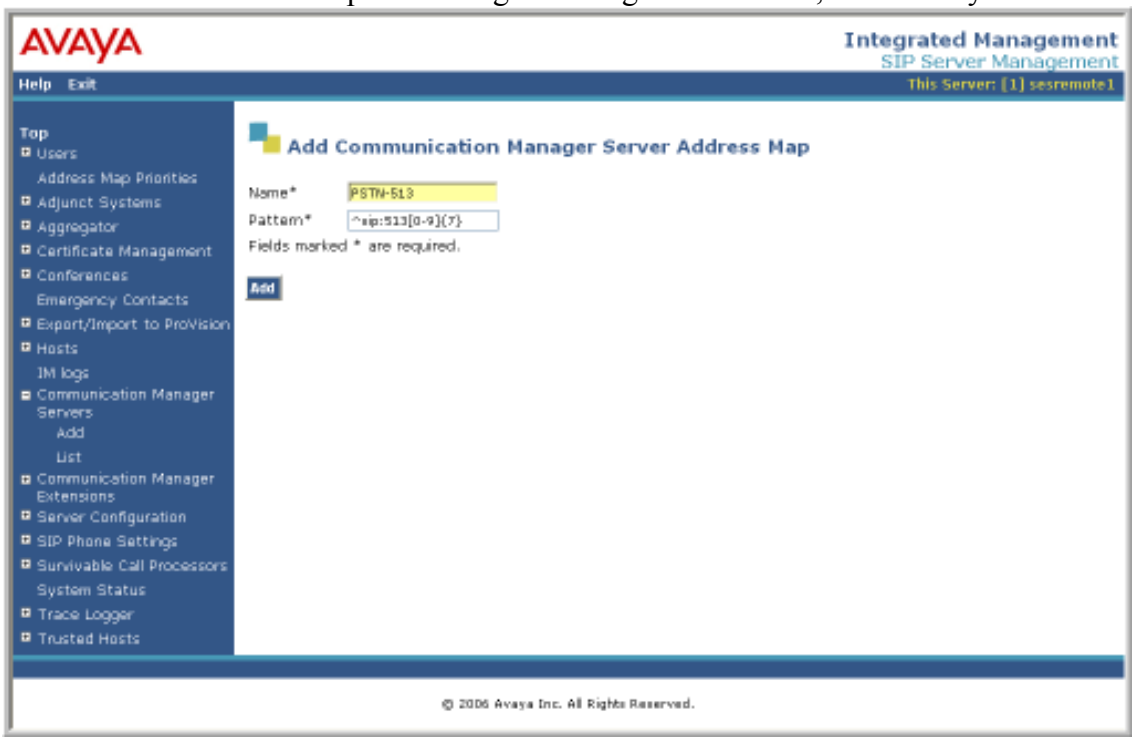

Figure 27: Communication Manager Server Address Map

After the **Communication Manager Server Address Map** is added, the first **Communication Manager Server Contact** is created automatically. For the address map added in **Figure 28**, the following contact was created:

```
sip:$(user)@2.160.183.202:5061;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 Communication Manager Server Address Map** screen appears as shown in **Figure 28**.

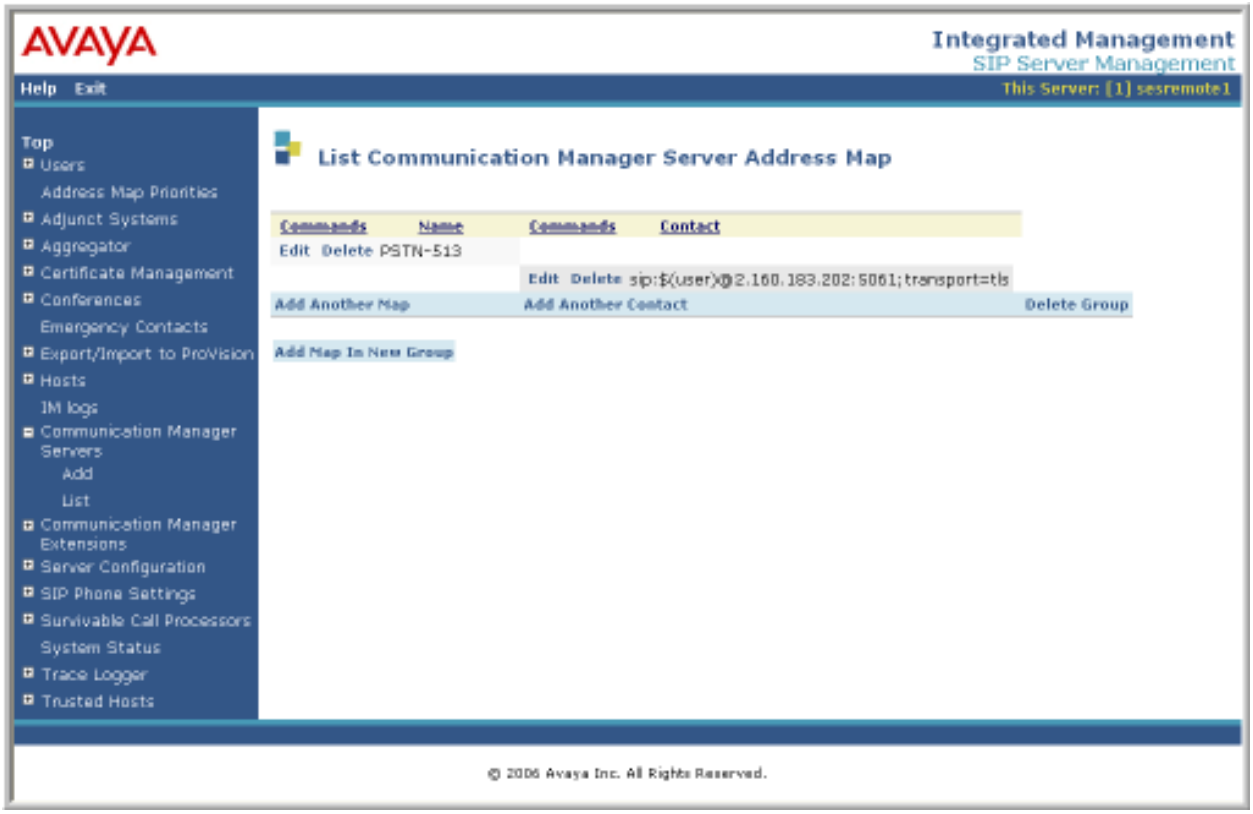

Figure 28: List Media Server Address Map

Add a user for each Avaya SIP telephone registering with 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 SIP Enablement Services server hosting the domain (*sipsp.avaya.com*) for this user. Enter the **First Name** and **Last Name** of the user. To associate a Communication Manager server 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. The **Add Communication Manager Extension** screen shown in **Figure 30** will be displayed after adding this user profile by clicking on the **Add** button.

Figure 29: Add User

In the **Add Communication Manager Extension** screen, enter the **Extension** configured on the media server, shown in **Figure 30**, for the previously added user. Usually, the media server extension and the user extension are the same (recommended). Select the **Communication Manager Server** assigned to this extension. Click the **Add** button.

Figure 30: Add Media Server Extension

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

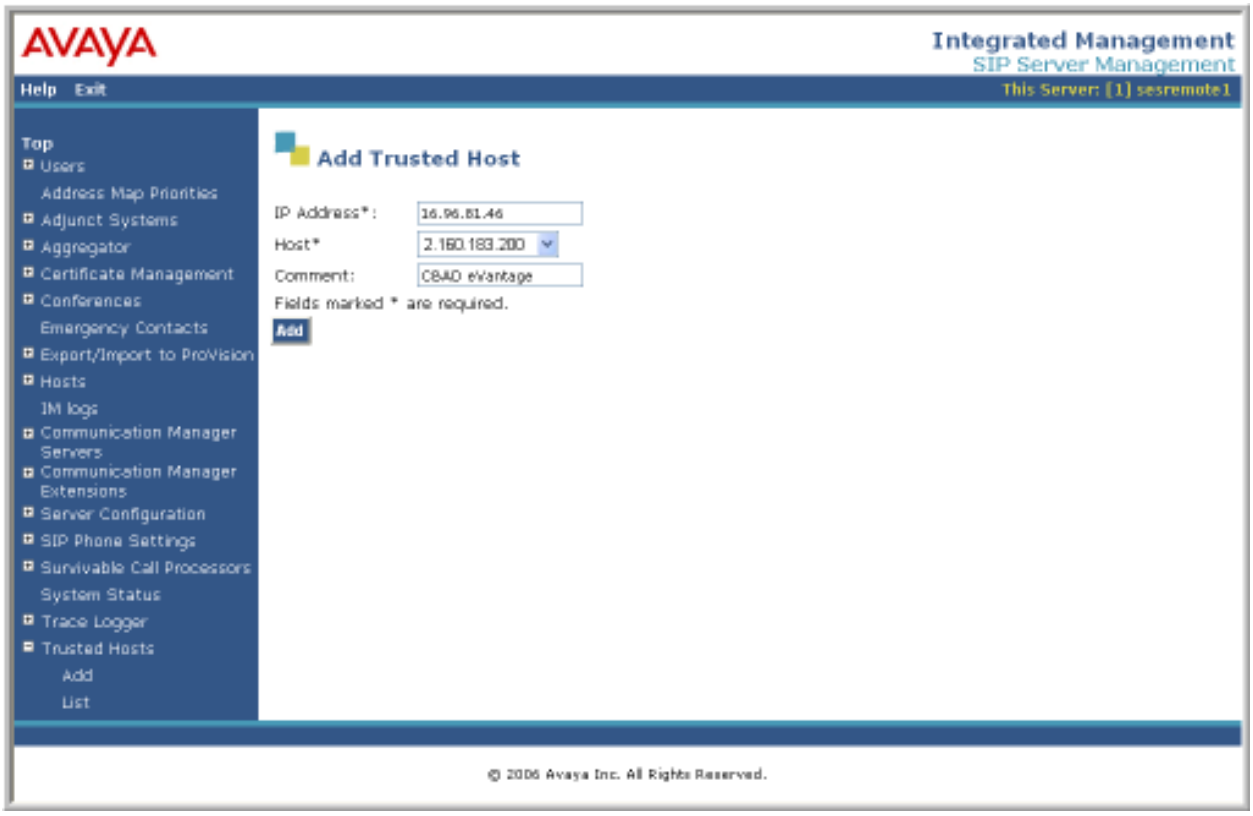

Figure 31: Add Trusted Host

6. Configure the Cincinnati Bell Any Distance eVantage Solution

To use Cincinnati Bell Any Distance (CBAD) eVantage solution, a customer must request service from Cincinnati Bell using their sales processes. Sales information for Cincinnati, Dayton, Ohio and Northern Kentucky can be reached at 1-888-CIN-BELL (246-2355). All other areas should call 1-317-816-5100, Option 1.

The following table contains the configuration information, coordinated with CBAD, which was used during the interoperability compliance testing to verify the *CBAD eVantage Solution*.

Feature	Test Configuration
Specify Codec(s) Required: ▪ G.711mu-law ▪ G.729A ▪ RFC2833 DTMF (required)	The network configuration described in these Application Notes was tested with the codecs (payload types) listed in the left column. **Note:** RFC2833 is required for shuffling SIP calls.
Define Dial Plan	10-digit dialing, directory assistance, toll-free, international, operator, and collect calls were supported by the test configuration.
Listed Directory Numbers provided by CBAD	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 513 were assigned to the SIP, H.323, digital, and analog endpoints in the enterprise network. In addition, these DID numbers will be sent as the CPN to the CBAD VoIP network for authentication.
CBAD provides Proxy IP Address	The IP address of the Cisco CUBE SIP proxy/SBC to reach the CBAD eVantage solution was 16.96.81.46.
Customer provides IP Address of SIP Enablement Services	The IP address of SIP Enablement Services in the enterprise network was 2.160.183.200. CBAD 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 SIP Enablement Services and CBAD eVantage Solution using UDP and port 5060.

7. General Test Approach and Test Results

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between Cincinnati Bell Any Distance eVantage solution and the Avaya SIP based network. This section covers the general test approach and the test results.

An enterprise site containing an Avaya SIP based network was connected using SIP trunking (via general purpose Internet services) to the CBAD eVantage solution. The SIP trunk was established between SIP Enablement Services and a Cisco CUBE. This allowed the enterprise site to access the PSTN network through the CBAD eVantage solution. The general test approach included the following:

- Incoming calls to the Avaya IP network from the PSTN routed through the CBAD VoIP network.
- Outgoing calls from the Avaya IP network to the PSTN routed through the CBAD 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, operator, and directory assistance calls.
- Voice calls using G.711 and G.729 codecs, including codec negotiation. For codec negotiation, the CBAD VoIP network will select its configured preferred codec for the call.
- DTMF transmission using RFC 2833.
- 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. See EC500 and call forwarding issues identified in the observations list.

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

The following observations were noted:

1. Cincinnati Bell does not support 0+ dialing from their services. Users will hear a recorded message that the number dialed is out of service. Users may dial 0 to reach an automated attendant.

2. eVantage does not support FAX capabilities across the SIP trunk to the PBX, but can provide FAX lines via traditional FXS connections.

3. Cincinnati Bell supports incoming toll free numbers by routing the toll free number to a specified DID number.

4. **EC500:** The EC500 feature (i.e., Extension to Cellular) applies to a user who can be reached at their Avaya desk phone or a cellular phone over the PSTN by dialing a single DID number. When a call is made to this DID number from the PSTN, the desk phone and cellular phone should ring simultaneously allowing the user to answer the call on either phone depending on their location. However, in this configuration, when an incoming PSTN call arrives to an Avaya desk phone with EC500 enabled, the outgoing EC500 call to the user's cellular phone over the PSTN is denied by the CBAD VoIP network. The outgoing call is denied because Avaya sends out the calling number of the PSTN user, which is unknown to the CBAD VoIP network and can't be authenticated. In this case, only the Avaya desk phone will ring since the outgoing EC500 call was denied. If the call originates from a local Avaya telephone, this issue does not occur because the CBAD VoIP network can authenticate the local Avaya user, if a DID number has been assigned to the user.

5. **Call Forwarding Off-Net:** This issue is similar to the EC500 issue described above in that an incoming PSTN call delivered to an Avaya station with Call Forwarding enabled to an off-net PSTN phone will be denied by the CBAD VoIP network because it won't be able to authenticate the calling number of the PSTN user sent by Avaya. In this case, the call will not be forwarded and the PSTN caller will hear "busy" tone. If the call originates from a local Avaya telephone, this issue does not occur because the CBAD VoIP network can authenticate the local Avaya user, if a DID number has been assigned to the user.
   Workaround: Enable the Special Application *(SA8972) – Overwrite Calling Identity* in the **system-parameters special-applications** form to overwrite the incoming calling party number (CPN) from the PSTN to the DID number of the local station. The **Overwrite Calling Identity** field on Page 4 of the outgoing SIP trunk group (e.g., trunk group 100 in this configuration) should also be set to (y)es.

6. **DTMF Tones to Avaya IP Phones:** When shuffling is enabled, IP phones will hear clicks. If the call is not shuffled, tones will be played out by the media processor board.

8. Verification Steps

This section provides verification steps that may be performed to verify that the H.323, digital and analog endpoints can place outbound and receive inbound calls through Cincinnati Bell Any Distance eVantage solution.

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 1 minute. 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 1 minute.

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 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*.

9. Conclusion

These Application Notes describe the configuration steps required to connect an enterprise site consisting of an Avaya SIP-based Network to the PSTN via the Cincinnati Bell Any Distance eVantage solution. The CBAD eVantage solution is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. The CBAD eVantage solution provides businesses a flexible, cost-saving alternative to traditional hardwired 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] *Administering Avaya Aura™ Communication Manager*, Document 03-300509, Issue 5.0, Release 5.2, May 2009,

[2] *SIP Support in Avaya Aura™ Communication Manager Running on the Avaya S8xxx Servers*, May 2009, Issue 9, Document Number 555-245-206.

[3] *Installing, Administering, Maintaining, and Upgrading Avaya Aura™ SIP Enablement Services*, May 2009, Issue 7, Document Number 03-600768.

[4] RFC 3261 *SIP: Session Initiation Protocol* http://www.ietf.org/

APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by the CBAD 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 CBAD eVantage Solution:

```
No.     Time        Source                Destination           Protocol Info
     21 9.844315     16.96.81.46           2.160.183.200           SIP/SDP   Request: INVITE
sip:5135551234@2.160.183.200:5060, with session description

Frame 21 (1165 bytes on wire, 1165 bytes captured)
Ethernet    II,    Src:    Cisco_6a:ef:e0    (00:07:eb:6a:ef:e0),    Dst:    Ibm_84:59:d3
(00:14:5e:84:59:d3)
Internet Protocol, Src: 16.96.81.46 (16.96.81.46), Dst: 2.160.183.200 (2.160.183.200)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
    Request-Line: INVITE sip:5135551234@2.160.183.200:5060 SIP/2.0
        Method: INVITE
        [Resent Packet: False]
    Message Header
        Via: SIP/2.0/UDP 16.96.81.46:5060;branch=z9hG4bK19E1571
            Transport: UDP
            Sent-by Address: 16.96.81.46
            Sent-by port: 5060
            Branch: z9hG4bK19E1571
        From: "REDBANK,NJ" <sip:7328523042@as.voip.fuse.net>;tag=91A4ABC-E88
            SIP Display info: "REDBANK,NJ"
            SIP from address: sip:7328523042@as.voip.fuse.net
            SIP tag: 91A4ABC-E88
        To: <sip:5135551234@2.160.183.200>
            SIP to address: sip:5135551234@2.160.183.200
        Date: Thu, 11 Jun 2009 14:22:22 GMT
        Call-ID: 1D392B02-55CA11DE-81B3D6BD-1647EF06@as.voip.fuse.net
        Supported: 100rel,timer,resource-priority,replaces
        Min-SE:  1800
        Cisco-Guid: 490125922-1439306206-2175719101-373812998
        User-Agent: Cisco-SIPGateway/IOS-12.x
        Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
        CSeq: 101 INVITE
            Sequence Number: 101
            Method: INVITE
        Timestamp: 1244730142
        Contact: <sip:7328523042@16.96.81.46:5060>
            Contact Binding: <sip:7328523042@16.96.81.46:5060>
                URI: <sip:7328523042@16.96.81.46:5060>
                    SIP contact address: sip:7328523042@16.96.81.46:5060
        Expires: 300
        Allow-Events: telephone-event
        Max-Forwards: 8
        Content-Type: application/sdp
        Content-Disposition: session;handling=required
        Content-Length: 291
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): CiscoSystemsSIP-GW-UserAgent 3383 8819 IN IP4
16.96.81.46
```

```
    Owner Username: CiscoSystemsSIP-GW-UserAgent
    Session ID: 3383
    Session Version: 8819
    Owner Network Type: IN
    Owner Address Type: IP4
    Owner Address: 16.96.81.46
Session Name (s): SIP Call
Connection Information (c): IN IP4 16.96.81.46
    Connection Network Type: IN
    Connection Address Type: IP4
    Connection Address: 16.96.81.46
Time Description, active time (t): 0 0
    Session Start Time: 0
    Session Stop Time: 0
Media Description, name and address (m): audio 16874 RTP/AVP 0 18 101
    Media Type: audio
    Media Port: 16874
    Media Proto: RTP/AVP
    Media Format: ITU-T G.711 PCMU
    Media Format: ITU-T G.729
    Media Format: 101
Connection Information (c): IN IP4 16.96.81.46
    Connection Network Type: IN
    Connection Address Type: IP4
    Connection Address: 16.96.81.46
Media Attribute (a): rtpmap:0 PCMU/8000
    Media Attribute Fieldname: rtpmap
    Media Format: 0
    MIME Type: PCMU
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 0-16
    Media Attribute Fieldname: fmtp
    Media Format: 101 [telephone-event]
    Media format specific parameters: 0-16
```

Sample SIP INVITE Message from SIP Enablement Services to CBAD eVantage Solution:

```
No.     Time        Source                Destination           Protocol Info
      1 0.000000    2.160.183.200        16.96.81.46       SIP/SDP  Request: INVITE
sip:7328523042@16.96.81.46, with session description

Frame 1 (1484 bytes on wire, 1484 bytes captured)
Ethernet II, Src: Ibm_84:59:d3 (00:14:5e:84:59:d3), Dst: Cisco_6a:ef:e0
(00:07:eb:6a:ef:e0)
Internet Protocol, Src: 2.160.183.200 (2.160.183.200), Dst: 16.96.81.46 (16.96.81.46)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
    Request-Line: INVITE sip:7328523042@16.96.81.46 SIP/2.0
        Method: INVITE
        [Resent Packet: False]
    Message Header
        Accept-Language: en
        Call-ID: 80c25eb69d64de1b2174a4c839a00
        CSeq: 1 INVITE
            Sequence Number: 1
            Method: INVITE
        From: "Lange Public"
<sip:5135551234@avremote.com:5061>;tag=80c25eb69d64de1b1174a4c839a00
            SIP Display info: "Lange Public"
            SIP from address: sip:5135551234@avremote.com:5061
            SIP tag: 80c25eb69d64de1b1174a4c839a00
        Record-Route:
<sip:2.160.183.200:5060;lr>,<sip:12.160.183.202:5061;lr;transport=tls>
        To: "7328523042" <sip:7328523042@16.96.81.46>
            SIP Display info: "7328523042"
            SIP to address: sip:7328523042@16.96.81.46
        Via: SIP/2.0/UDP
2.160.183.200:5060;branch=z9hG4bK8383830303033636367461.0,SIP/2.0/TLS
12.160.183.202;psrrposn=2;received=12.160.183.202;branch=z9hG4bK80c25eb69d64de1b3174a4
c839a00
            Transport: UDP
            Sent-by Address: 2.160.183.200
            Sent-by port: 5060
            Branch: z9hG4bK8383830303033636367461.0,SIP/2.0/TLS
        Content-Length: 214
        Content-Type: application/sdp
        Contact: "Lange Public" <sip:5135551234@12.160.183.202:5061;transport=tls>
            Contact Binding: "Lange Public"
<sip:5135551234@12.160.183.202:5061;transport=tls>
                URI: "Lange Public" <sip:5135551234@12.160.183.202:5061;transport=tls>
                    SIP Display info: "Lange Public"
                    SIP contact address: sip:5135551234@12.160.183.202:5061
        Max-Forwards: 68
        User-Agent: Avaya CM/R015x.02.0.947.3
        Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS,INFO,PUBLISH
        Supported: timer,replaces,join,histinfo,100rel
        Alert-Info: <cid:internal@16.96.81.46>;avaya-cm-alert-type=internal
        Min-SE: 1800
        Session-Expires: 1800;refresher=uac
        P-Asserted-Identity: "Lange Public" <sip:5135551234@avremote.com:5061>
        P-Charging-Vector: icid-value="AAS:342-b65ec2801de649d4c4a17b09a83"
        History-Info: <sip:7328523042@16.96.81.46>;index=1,"7328523042"
<sip:7328523042@16.96.81.46>;index=1.1
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
```

```
Owner/Creator, Session Id (o): - 1 1 IN IP4 12.160.183.202
    Owner Username: -
    Session ID: 1
    Session Version: 1
    Owner Network Type: IN
    Owner Address Type: IP4
    Owner Address: 12.160.183.202
Session Name (s): -
Connection Information (c): IN IP4 12.160.183.203
    Connection Network Type: IN
    Connection Address Type: IP4
    Connection Address: 12.160.183.203
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 2456 RTP/AVP 0 18 101
    Media Type: audio
    Media Port: 2456
    Media Proto: RTP/AVP
    Media Format: ITU-T G.711 PCMU
    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: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
```

APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within 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.