Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 5.2.1 and Avaya Aura® SIP Enablement Services 5.2.1 with Sprint IP Toll Free SIP Trunking Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager and Avaya Aura® SIP Enablement Services with Sprint IP Toll Free SIP Trunk service.

Avaya Aura® SIP Enablement Services is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® SIP Enablement Services. Avaya Aura® SIP Enablement Services is the point of connection between Avaya Aura® Communication Manager and the Sprint IP Toll Free service.

The Sprint IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks.

Sprint is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through 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 required for configuring Avaya Aura® SIP Enablement Services and Avaya Aura® Communication Manager with Sprint IP Toll Free SIP trunk service.

Avaya Aura® SIP Enablement Services is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® SIP Enablement Services. Avaya Aura® SIP Enablement Services is the point of connection between Avaya Aura® Communication Manager and the Sprint IP Toll Free service.

The Sprint IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks. Customers using this Avaya telephony solution with Sprint IP Toll Free SIP Trunk Service are able to receive PSTN calls via a dedicated broadband Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

2. General Test Approach and Test Results

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

The test environment consisted of:

- A simulated enterprise with Avaya Aura® SIP Enablement Services, Avaya Aura® Communication Manager, Avaya phones and fax machines, and Avaya Modular Messaging.
- A laboratory version of the Sprint IP Toll Free service, to which the simulated enterprise was connected via an IPSec VPN connection that emulated the Sprint MPLS network.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound and redirected call flows between Avaya Aura® SIP Enablement Services, Avaya Aura® Communication Manager and the Sprint IP Toll Free service.

The compliance testing was based on a test plan provided by Sprint and specific sections of the standard Avaya SIP trunk test plan, for the functionality required for certification as a supported solution. Calls were made from the PSTN across the Sprint network to the Avaya system at the enterprise. The following features were tested as part of this effort:

- Inbound IP Toll Free SIP trunking.

- T.38 Faxing.
- Passing of DTMF events and their recognition by navigating automated menus.
- IP PBX and Sprint IP Toll Free service features such as hold and resume, call forwarding, conference and transfer.

2.2. Test Results

The main test objectives were to verify the following features and functionality:

- Inbound Sprint IP Toll Free calls to Communication Manager phones, agents and VDNs/Vectors.
- Two-way talk path establishment between the PSTN and Communication Manager endpoints via the Sprint Toll Free service.
- Basic supplementary telephony features such as hold and resume, call forwarding, conference and transfer.
- G.729A and G.711MU codecs.
- T.38 fax calls.
- DTMF tone transmission using RFC 2833.
- Inbound Sprint IP Toll Free service calls to Communication Manager that are directly routed to stations, and if unanswered, are either redirected or can be covered to Avaya Modular Messaging.
- Long duration calls.

The test objectives stated in **Section 2.1** were verified.

2.3. Observations and Limitations

During the compliance testing the following observations and limitations were documented:

- When Caller ID Block was invoked, the same 10 digit number was being displayed for all calls that requested privacy. This was determined by Sprint to be a condition of the lab setup, and should not adversely affect service in production.
- There was only the ability to complete one call at a time per IPTF DID that had been assigned, therefore, the compliance test was not able to simulate multiple, simultaneous calls to the same toll free number as would be seen in a real-world call center setup.

2.4. Support

Sprint customers may obtain information for Sprint IP Toll Free service by going to www.sprint.com or for technical support contact a Sprint Customer Care representative.

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on http://support.avaya.com) to directly access specific support and consultation services based upon their Avaya support agreements.

3. Reference Configuration

The reference configuration used in these Application Notes is shown in **Figure 1** and consists of several components:

- Avaya Aura® SIP Enablement Services (SES) runs on an Avaya S8500 server. SES serves as a SIP proxy between Communication Manager and the Sprint IP Toll Free service. SES also provides registrar services to SIP phones in the enterprise.
- Avaya Aura® Communication Manager provides the Enterprise Voice communications services. In this sample configuration, Avaya Aura® Communication Manager runs on an Avaya S8730 Server. This solution is extensible to other Avaya S8xxx Servers as well.
- The Avaya Media Gateway provides the physical interfaces and resources for Avaya Aura® Communication Manager. In the reference configuration, an Avaya G650 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya "desk" phones are represented with Avaya analog, 2420 Digital, 4621 SW IP and 9630 Series IP Telephones running H.323 or SIP software, as well as an Avaya one-X® Communicator softphone.
- An existing Avaya Modular Messaging system provides the corporate voice messaging capabilities in the reference configuration. The provisioning of Modular Messaging is beyond the scope of this document.
- Inbound calls were placed from the PSTN via the Sprint IP Toll Free service, to the Avaya Aura® SIP Enablement Services, which routed the call to Avaya Aura® Communication Manager. Avaya Aura® Communication Manager terminated the call to the appropriate agent/phone or fax extension. The H.323 phones on the enterprise side registered to the Avaya Aura® Communication Manager CLAN IP address on the Avaya G650 media gateway. The SIP phones registered to Avaya Aura® SIP Enablement Services.

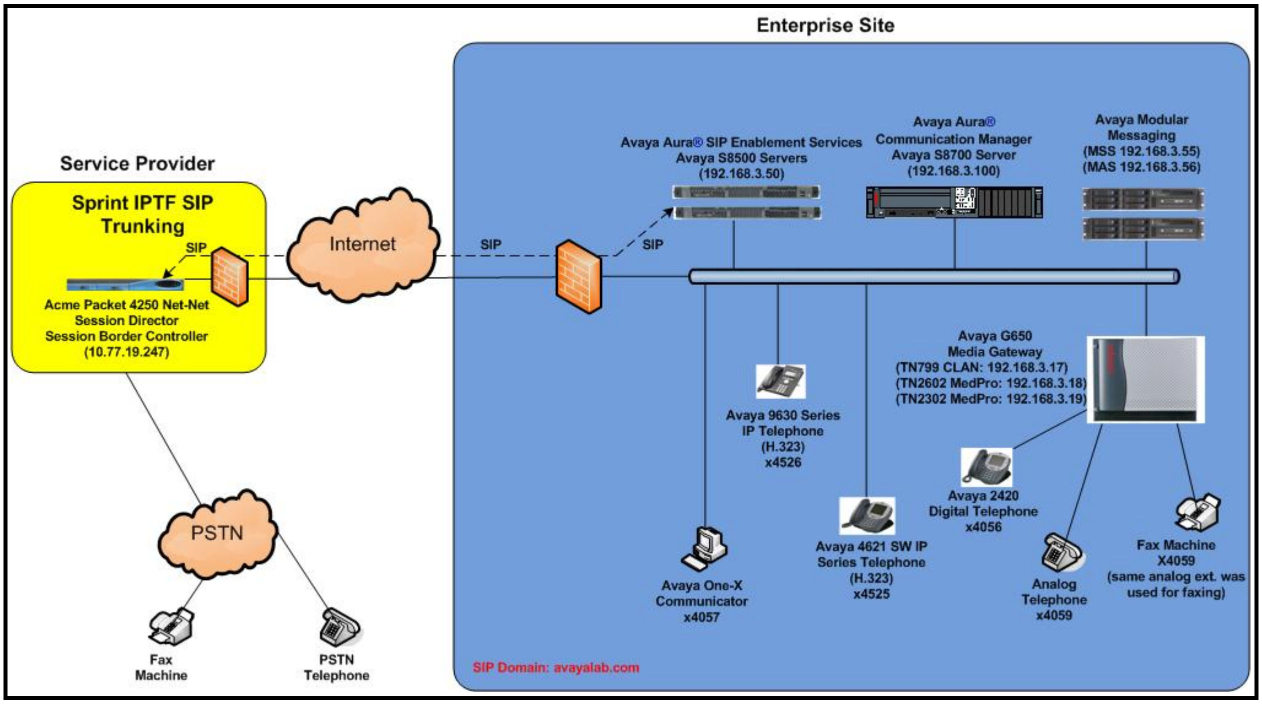

Figure 1: Reference configuration

3.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the reference configuration described in these Application Notes, and are **for illustrative purposes only**. Customers must obtain and use the specific values for their own specific configurations.

Note - The Sprint IP Toll Free service Session Border Controller IP address and IPTF DID numbers, (destination digits specified in the SIP Request URIs sent by the Sprint IP Toll Free service) are shown in this document as examples. Sprint will provide actual IP addresses and DID numbers as part of the IP Toll Free provisioning process.

Component	Illustrative Value in these Application Notes
Avaya Aura® SIP Enablement Services	
IP Address	192.168.3.50
Avaya Aura® Communication Manager	
C-LAN IP Address	192.168.3.17
VDN	4020
Skill (Hunt Group)	1, 2, 3
Agent Extensions	5001, 5002, 5003, 5004
Hunt Group Extensions	4901 (Skill 1), 4902 (Skill 2), 4903 (Supergroup / Skill 3)
Phone Extensions	4056, 4059, 4525, 4526, 4529
Voice Messaging Pilot Extension	7999
Avaya Modular Messaging	
Messaging Application Server (MAS) IP Address	192.168.3.56
Message Store Server (MSS) IP Address	192.168.3.55
Sprint IP Toll Free Service	
Acme Packet SBC IP Address	10.77.19.247
IPTF DIDs Assigned by Sprint	855-551-1818 855-551-1819 855-551-1820 855-551-1821

Table 1: Illustrative Values Used in these Application Notes

4. Equipment and Software Validated

The following equipment and software was used for the reference configuration described in these Application Notes.

Component	Version
Avaya S8510 Server	Avaya Aura® SIP Enablement Services 5.2.1 SP6c (SES-5.2.1.0-016.4-SP6c)
Avaya S8730 Server	Avaya Aura® Communication Manager 5.2.1 SP10 (R015x.02.1.016.4-19191)
Avaya G650 Media Gateway	
CONTROL-LAN TN799DP IP	HW01 FW024
VAL-ANNOUNCEMENT TN2501AP	HW03 FW018
IP MEDIA PROCESSOR TN2602AP	HW08 FW044
IP MEDIA PROCESSOR TN2302AP	HW18 FW110
DIGITAL LINE TN2224	Vintage 000007
ANALOG LINE TN746B	Vintage 000004
Avaya 9630 IP Telephone	H.323 Version S3.110b (ha96xxua3_11.bin)
Avaya 9630 IP Telephone	SIP Version 2.6.0
Avaya one-X® Communicator	6.0.1.16-SP1-25226
Avaya 4621SW IP Telephone	H323 Version 2.9.1 (a10d01b2_9_1.bin)
Avaya Analog phone	N/A
Fax device	Okidata Okifax
Sprint IPTF Service	
Sprint IP Toll Free Service Acme Packet Net-Net Session Director	SCX6.2.0

Table 2: Equipment and Software Versions

5. Avaya Aura® Communication Manager

This section describes the administration steps for Avaya Aura® Communication Manager in support of SIP trunking integration with the Sprint IP Toll Free service. The steps are performed from the Communication Manager System Access Terminal (SAT) interface. These Application Notes assume that basic Communication Manager Administration, including stations, C-LAN and Media Processor boards, SIP phone signaling/trunk group(s), etc., has already been performed. Refer to **[1]**, **[2]**, and **[3]** for further details if necessary.

Note – In the following sections, only the **highlighted** parameters are applicable to these Application Notes. Other parameters shown should be considered informational.

5.1. System Parameters

This section reviews the Communication Manager licenses and features that are required for the sample configuration described in these Application Notes. For required licenses that are not enabled in the steps that follow, contact an authorized Avaya account representative to obtain the licenses.

1. Enter the **display system-parameters customer-options** command. On Page 2 of the **system-parameters customer-options** form, verify that the **Maximum Administered SIP Trunks** number is sufficient for the number of expected SIP trunks.

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

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 1000  30
           Maximum Concurrently Registered IP Stations: 18000 3
             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: 1000  353
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 128   1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   1
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

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

Figure 4: System-Parameters Customer-Options Form – Page 2

2. On Page 4 of the **system-parameters customer-options** form, verify that the bolded fields in the following screenshots are set to "**y**".

```
display system-parameters customer-options                    Page   4 of  10
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                             IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                       ISDN Feature Plus? n
                  Enhanced EC500? y     ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                         ISDN-BRI Trunks? n
       Enterprise Wide Licensing? n                                ISDN-PRI? y
              ESS Administration? n              Local Survivable Processor? n
          Extended Cvg/Fwd Admin? n                    Malicious Call Trace? y
     External Device Alarm Admin? n                Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n   Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? n                Multifrequency Signaling? y
      Global Call Classification? n        Multimedia Call Handling (Basic)? n
             Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? n
 Hospitality (G3V3 Enhancements)? n             Multimedia IP SIP Trunking? n
                       IP Trunks? y

           IP Attendant Consoles? N

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

Figure 5: System-Parameters Customer-Options Form – Page 4

5.2. Dial Plan

This section briefly describes the dial plan requirements and feature access codes for the sample configuration described in these Application Notes. Enter the **change dialplan analysis** command to provision the dial plan as shown below.

- 3-digit dial access codes (indicated with a **Call Type** of "**dac**") beginning with the digit "**1**" – Trunk Access Codes (TACs) defined for trunk groups in this sample configuration conform to this format.
- 4-digit extensions with a **Call Type** of "**ext**" beginning with the digits "**4**", "**5**" or "**7**"– Local extensions for Communication Manager stations, agents, voicemail access, etc. in this sample configuration conform to this format.
- Feature access codes beginning with "*" or "**#**" to access various features available on the system, and an entry for a feature access code of "**9**" to access an outside line.

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

       Dialed    Total  Call    Dialed    Total  Call     Dialed    Total  Call
       String    Length Type    String    Length Type     String    Length Type
     1             3    dac
     4             4    ext
     5             4    ext
```

```
      7            4    ext
      9            1    fac
      *            3    fac
      #            3    fac
```

Figure 6: Dialplan Analysis Form

5.3. IP Network Parameters

For simplicity in this sample configuration, all Communication Manager elements, e.g., stations, C-LAN and MedPro boards, etc., within the Avaya site are assigned to a single IP network region and all internal calls use a single IP codec set. This section describes the steps required for administering IP network regions and IP codec sets, for both the enterprise and the Sprint IP Toll Free service.

1. Enter the **change ip-codec-set *x*** command, where ***x*** is the number of an IP codec set used only for internal calls and configure as follows:

```
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:
```

Figure 7: IP-Codec-Set Form for Internal Calls – Page 1

- On Page 2 of the **ip-codec-set** form, set **FAX Mode** 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         US                   3
    Clear-channel   n                    0
```

Figure 8: IP-Codec-Set Form for Internal Calls – Page 2

2. Enter the **change ip-codec-set *x*** command, where ***x*** is the number of an unused IP codec set. This IP codec set will be used for inbound Sprint IP Toll Free calls. On Page 1 of the **ip-codec-set** form, provision the codecs in the order shown in below.

```
change ip-codec-set 3                                          Page   1 of   2

                          IP Codec Set

    Codec Set: 3

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

Figure 9: IP-Codec-Set Form for Inbound Calls – Page 1

- On Page 2 of the **ip-codec-set** form, set **FAX Mode** to "**t.38-standard**".

```
change ip-codec-set 3                                          Page   2 of   2

                          IP Codec Set

                              Allow Direct-IP Multimedia? n


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

Figure 10: IP-Codec-Set Form for Inbound Calls – Page 2

3. Enter the **change ip-network-region *x***, where ***x*** is the number of an unused IP network region. This IP network region will be used to represent the Sprint IP Toll Free Service. On Page 1 of the **ip-network-region** form:
   - Set the **Authoritative Domain** field to the IP address or FQDN of the Sprint IPTF SBC. For the compliance test this field was set to the IP address of the Sprint IPTF SBC, **10.77.19.247**.
   - **Codec Set** – Set to the codec set configured in **Step 2**
   - **Intra** and **Inter-region IP-IP Direct Audio** – Set both to **yes** to support audio shuffling which allows the endpoints to communicate directly with each other.

```
change ip-network-region 3                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 3
Location:          Authoritative Domain: 10.77.19.247
    Name: Sprint IPTF
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 3              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                        RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Figure 11: IP-Network-Region Form for Sprint IP Toll Free service – Page 1

On Page 3 of the **ip-network-region** form, provision the following:
- **codec set** – Set to the codec set configured in **Step 2**
- **direct WAN** – Set to **y**

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

 Source Region:  3     Inter Network Region Connection Management

 dst codec direct   WAN-BW-limits   Video      Intervening   Dyn
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions       CAC IGAR AGL
 1    3     y    NoLimit                                          n   all
 3    3                                                               all
```

Figure 12: IP-Network-Region Form for Sprint IP Toll Free service – Page 3

4. Enter the **change ip-network-region *x***, where ***x*** is the number of an IP network region administered for local Communication Manager Elements within the Avaya site.
   On Page 1 of the **ip-network-region** form:
   - Set the **Authoritative Domain** field to the domain name already configured in SES. The domain of **avayalab2.com** was used for the compliance test.
   - **Codec Set** – Set to the codec set configured in **Step 1**
   - Set **Intra-region** and **Inter-region IP-IP Direct Audio** to **yes** to support audio shuffling which allows the endpoints to communicate directly with each other.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avayalab2.com
    Name: Enterprise
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Figure 13: IP-Network-Region Form for Local Communication Manager Elements – Page 1

On Page 3 of the **ip-network-region** form, provision the following:
- **codec set** – Set to the codec set administered in **Step 1**
- **direct WAN** – Set to **y**

```
change ip-network-region 1                                      Page   3 of  19

 Source Region:  1     Inter Network Region Connection Management

 dst codec direct   WAN-BW-limits   Video      Intervening    Dyn
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions       CAC IGAR AGL
 1    1     y    NoLimit                                          n   all
 2
 3    3     y    NoLimit                                          n   all
```

Figure 14: IP-Network-Region Form for Local Communication Manager Elements – Page 3

5. Enter the **list node-names all** command, and note the node names and IP addresses of the SES server as well as the C-LAN board for the G650 gateway that connects to Communication Manager.

```
list node-names all

                        NODE NAMES

Type     Name               IP Address
IP       Acme               192.168.3.3
IP       Gateway001         192.168.3.1
IP       MM                 192.168.3.56
IP       clan1              192.168.3.17
IP       default            0.0.0.0
IP       medpro1            192.168.3.18
IP       medpro2            192.168.3.19
IP       procr               .  .  .
IP       ses                192.168.3.50
```

Figure 15: Node-Names Form

5.4. Inbound Calls

In this sample configuration, since the Sprint IP Toll Free service sends 10-digits on inbound calls, the administration steps that follow in this section reflect that requirement. In an actual deployment, this may be different.

5.4.1. Inbound IP Toll Free Calls

This section describes the steps for administering the routing of inbound calls from the PSTN.

1. Enter the **add signaling-group *x*** command, where ***x*** is the number of an unused signaling group and configure as follows:
   - Group Type – Set to **sip**
   - **Transport Method** – Set to **tls** for this testing. For security reasons it is recommended to use TLS for deployment in production environments.
   - **Near-end Node Name** - Set to the CLAN node name configured in **Section 5.3**, **Step 5**
   - **Far-end Node Name** – Set to the SES node name configured in **Section 5.3**, **Step 5**
   - Near-end and Far-end Listen Ports are set to the default value of **5061** for **Transport Method** of **tls**
   - **Far-end Domain** – Set to the IP address or FQDN of the Sprint IPTF SBC.
   - **DTMF over IP** - Set to "**rtp-payload**" to enable Avaya Aura® Communication Manager to use DTMF according to RFC 2833
   - **Direct IP-IP Audio Connections** – Set to **y**, indicating that the RTP paths should be optimized to reduce the use of media processing resources when possible

```
 add signaling-group 4                                         Page   1 of   1
                                SIGNALING GROUP

 Group Number: 4              Group Type: sip
                        Transport Method: tls
  IMS Enabled? n


   Near-end Node Name: clan1                 Far-end Node Name: ses
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 3
Far-end Domain: 10.77.19.247

                                           Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                RFC 3389 Comfort Noise? 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                     Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

Figure 16: Signaling-Group Form for Sprint IP Toll Free Calls

2. Enter the **add trunk-group *x*** command, where ***x*** is the number of an unused trunk group and configure as follows:
   - **Direction** – Set to **two-way**. This allows a single trunk to be used for inbound calls and calls that need to be redirected back out of the enterprise to the PSTN.
   - **TAC** – Set to any value as per the dial plan
   - **Service Type** – Set to **public-ntwrk**
   - **Signaling Group** – Set to the number of the signaling group administered in **Step 1**
   - **Number of Members** – Set to a value large enough to accommodate the call volume. For the compliance testing, this value was set to **14**.

```
add trunk-group 4                                               Page   1 of  21
                                TRUNK GROUP

Group Number: 4                   Group Type: sip          CDR Reports: y
  Group Name: Sprint IPTF                COR: 1        TN: 1        TAC: 104
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                     Signaling Group: 4
                                                   Number of Members: 14
```

Figure 17: Trunk-Group Form for Sprint IP Toll Free Calls – Page 1

3. Enter the **change inc-call-handling-trmt trunk-group *x*** command, where ***x*** is the number of the trunk group administered in **Step 2** above. In the **inc-call-handling-trmt trunk-group** form, provision an entry for an IPTF number in the SIP INVITE message as follows:
   - **Called Len** – Enter the total number of digits in the IPTF number
   - **Called Number** – Enter enough leading digits to uniquely match the IPTF range
   - **Del** and **Insert** – If necessary, enter the number of leading digits that need to be deleted from the IPTF number, and the specific leading digits that need to be prefixed to the IPTF number (after any deletion is performed), respectively, in order to match a local Communication Manager VDN / extension range.

   In this sample configuration, Sprint IP Toll Free service sends 10-digits in the SIP INVITE (**8555511819**) on inbound IPTF calls. Thus, the entry in the figure below matches this number and deletes all 10 digits. Then it inserts **4020** to match the local Communication Manager VDN. VDNs are described in **Section 5.5**.
   Provision as many entries as necessary to cover all expected IPTF numbers sent by Sprint IP Toll Free service.

```
change inc-call-handling-trmt trunk-group 4                      Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called          Del Insert
 Feature        Len      Number
 public-ntwrk    10      8555511819       10  4020
```

Figure 18: Inc-Call-Handling-Trmt Trunk-Group Form for Sprint IP Toll Free Calls

5.4.2. Calling Party Number

The calling party numbers sent on outbound calls can be specified in the **public-unknown-numbering** form. These numbers are displayed for Caller ID for any connected calls or redirected outbound calls. Configure an entry in the **public-unknown-numbering** form as follows:

- **Ext Len** – Enter the total number of digits in the local extension range.
- **Ext Code –** Enter enough leading digits to identify the local extension range.
- **Trk Grp(s)** – Enter the number of the trunk group administered in **Section 5.4.1**, **Step 2**.
- **CPN Prefix –** If necessary, enter enough prefix digits to form the desired connected party number.
- **CPN Len** – Enter the total length of the connected party number to be sent.

Provision as many entries as necessary to cover all local Extension ranges.

```
Change public-unknown-numbering 4                                 Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext              Trk       CPN              CPN
Len Code             Grp(s)    Prefix           Len
                                                     Total Administered: 1
 4  4059             4         7205559453       10      Maximum Entries: 9999
 4  4056             4         7205559454       10
 4  4525             4         7205559455       10
 4  4526             4         7205559456       10
```

Figure 19: Public-Unknown-Numbering Form

5.5. Optional Features

The reference configuration uses Hunt Groups, Vectors, and Vector Directory Numbers (VDNs), to provide additional functionality during testing:

- Hunt Group 1 (4901) – Hunt Group for Agents in Skill Group 1
- Hunt Group 2 (4902) – Hunt Group for Agents in Skill Group 2
- Hunt Group 3 (4903) – Hunt Group for Agents in the Supergroup
- Hunt Group 99 – Modular Messaging coverage for Communication Manager extensions
- VDN 4020/Vector 2 – VDN and vector used to redirect unanswered calls back out to Sprint via a SIP 302 Redirect message (302 Moved Temporarily)
- VDN 4020/Vector 4 – VDN and vector used to redirect answered calls back out to Sprint via the SIP REFER method. This is used for blind and consultative transfers, as well as, forwarded calls and automated attendant calls.

Note - The administration of Communication Manager Call Center elements – hunt groups, vectors, and Vector Directory Numbers (VDNs) are beyond the scope of these Application Notes. Additional licensing may be required for some of these features. Refer to **[1]**, **[2]**, **[7]**, and **[8]** for further details if necessary. The samples that follow are provided for reference purposes only.

5.5.1. Hunt Group for Station Coverage to Modular Messaging

Hunt group 99 is used in the reference configuration to verify the Send-All-Calls functionality. The hunt group (e.g. **99**) is defined with the 4 digit Modular Messaging pilot number (e.g. **7999** in **Figure 21**). The hunt group is associated with a coverage path (e.g.**h99** in **Figure 22**) and the coverage path is assigned to a station/agent.

```
display hunt-group 99                                          Page   1 of  60
                                  HUNT GROUP

            Group Number: 99                                ACD? n
              Group Name: mm                              Queue? n
         Group Extension: 7999                           Vector? n
              Group Type: ucd-mia                  Coverage Path:
                      TN: 1           Night Service Destination:
                     COR: 1                     MM Early Answer? n
           Security Code:                Local Agent Preference? n
 ISDN/SIP Caller Display: mbr-name
```

Figure 20: Hunt Group Form – Page 1

```
display hunt-group 99                                          Page   2 of  60
                                  HUNT GROUP

                      Message Center: sip-adjunct

     Voice Mail Number         Voice Mail Handle         Routing Digits
                                                  (e.g., AAR/ARS Access Code)
     7999                      voicemail                 *88
```

Figure 21: Hunt Group Form – Page 2

```
display coverage path 1                                       Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 1
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage
COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?             n               n
               Busy?             y               y
        Don't Answer?            y               y        Number of Rings: 2
                All?             n               n
 DND/SAC/Goto Cover?             y               y
   Holiday Coverage?             n               n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h99           Rng:    Point2:
  Point3:                       Point4:
  Point5:                       Point6:
```

Figure 22: Coverage Path Form

5.5.2. Call Center Provisioning

For provisioning the call center functionality, verify that the call center parameters are enabled as shown below. Verify that an agent login id is created with an appropriate skill. Verify the skill (hunt group) for that agent is in place. Make sure that a VDN as per the dial plan is in place along with the vector which lists the steps to be executed when an inbound call is received from Sprint IP Toll Free service.

In the reference configuration below, an inbound call from Sprint IP Toll Free service is handled using the incoming call handling treatment configured in **Section 5.4.1**, **Step 3**, using the VDN 4020 (**Figure 28**) which routes the call to Vector 4 (**Figure 29**) and based upon the digit input by the caller, the call is directed to an appropriate skill. Skill 1 (**Figure 30**) is shown for reference purposes and additional skills can be similarly added.

```
display system-parameters customer-options                      Page   6 of  10
                         CALL CENTER OPTIONAL FEATURES

                          Call Center Release: 5.0

                                ACD? y                          Reason Codes? n
                        BCMS (Basic)? y              Service Level Maximizer? n
          BCMS/VuStats Service Level? n            Service Observing (Basic)? y
  BSR Local Treatment for IP & ISDN? n     Service Observing (Remote/By FAC)? n
                  Business Advocate? n             Service Observing (VDNs)? n
                    Call Work Codes? n                            Timed ACW? n
      DTMF Feedback Signals For VRU? n                     Vectoring (Basic)? y
                   Dynamic Advocate? n                 Vectoring (Prompting)? y
       Expert Agent Selection (EAS)? y             Vectoring (G3V4 Enhanced)? y
                            EAS-PHD? y              Vectoring (3.0 Enhanced)? y
                   Forced ACD Calls? n     Vectoring (ANI/II-Digits Routing)? y
               Least Occupied Agent? y     Vectoring (G3V4 Advanced Routing)? y
          Lookahead Interflow (LAI)? n                     Vectoring (CINFO)? y
Multiple Call Handling (On Request)? n      Vectoring (Best Service Routing)? y
    Multiple Call Handling (Forced)? n                  Vectoring (Holidays)? y
  PASTE (Display PBX Data on Phone)? n                 Vectoring (Variables)? Y
        (NOTE: You must logoff & login to effect the permission changes.)
```

Figure 23: Call Center Optional Features Form

```
display agent-loginID 5001                                      Page   1 of   2
                              AGENT LOGINID

                Login ID: 5001                                      AAS? n
                    Name: Agent Skill 1                           AUDIX? n
                      TN: 1                               LWC Reception: spe
                     COR: 1                       LWC Log External Calls? n
           Coverage Path: 1                     AUDIX Name for Messaging:
           Security Code: 1234
                                          LoginID for ISDN/SIP Display? n
                                                                Password: 1234
                                                 Password (enter again): 1234
                                                             Auto Answer: none
                                                      MIA Across Skills: system
                                              ACW Agent Considered Idle: system
                                              Aux Work Reason Code Type: system
                                                Logout Reason Code Type: system
                       Maximum time agent in ACW before logout (sec): system
                                              Forced Agent Logout Time:   :
     WARNING:  Agent must log in again before changes take effect
```

Figure 24: Agent Form – Page 1

```
display agent-loginID 5001                                      Page   2 of   2
                                 AGENT LOGINID
      Direct Agent Skill:                                  Service Objective? n
Call Handling Preference: skill-level                  Local Call Preference? n

    SN    RL SL           SN    RL SL           SN    RL SL           SN    RL SL
 1: 1        1       16:                 31:                 46:
 2: 3        1       17:                 32:                 47:
 3:                  18:                 33:                 48:
```

Figure 25: Agent Form – Page 2

```
display hunt-group 1                                            Page   1 of   3
                                  HUNT GROUP

            Group Number: 1                                ACD? y
              Group Name: ACD Skill 1                    Queue? y
         Group Extension: 4901                          Vector? y
              Group Type: ucd-mia
                      TN: 1
                     COR: 1                     MM Early Answer? n
           Security Code:                Local Agent Preference? n
 ISDN/SIP Caller Display:

             Queue Limit: unlimited
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
```

Figure 26: Skill (Hunt Group) Form – Page 1

```
display hunt-group 1                                            Page   2 of   3
                                  HUNT GROUP

                    Skill? y      Expected Call Handling Time (sec): 180
                      AAS? n
                 Measured: internal
     Supervisor Extension:

      Controlling Adjunct: none

Interruptible Aux Threshold: none
                                 Redirect on No Answer (rings):
                                               Redirect to VDN:
                   Forced Entry of Stroke Counts or Call Work Codes? n
```

Figure 27: Skill (Hunt Group) Form – Page 2

```
display vdn 4020                                                Page   1 of   3
                              VECTOR DIRECTORY NUMBER

                              Extension: 4020
                                  Name*: Sprint IPTF
                            Destination: Vector Number        2

                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: internal

                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:

* Follows VDN Override Rules
```

Figure 28: Select Skill VDN

```
display vector 2                                                   Page   1 of   6
                                 CALL VECTOR

    Number: 2                Name: Inbound Agents
                                             Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 collect      1    digits after announcement 4000     for none
03 goto step    8             if digits          =      1
04 goto step    10            if digits          =      2
05 goto step    12            if digits          =      3
06 goto step    14            if digits          =      4
07 stop
08 queue-to     skill 1    pri h
09 stop
10 queue-to     skill 2    pri h
11 stop
12 queue-to     skill 3    pri h
13 stop
14 route-to     number ~r13035551682    with cov n if unconditionally
```

Figure 29: RouteToSkill Vector

6. Avaya Modular Messaging

In this sample configuration, Avaya Modular Messaging is provisioned for Multi-Site mode. Multi-Site mode allows Avaya Modular Messaging to serve subscribers in multiple locations. The administration for Modular Messaging is beyond the scope of these Application Notes. Refer to **[10]** and **[11]** for further details.

7. Configure Avaya Aura® SIP Enablement Services

This section describes the administration steps for Avaya Aura® SIP Enablement Services (SES) in support of SIP trunking with the Sprint IP Toll Free service. These Application Notes assume that the necessary SES licenses have been installed and basic SES administration has already been performed. Refer to **[3]**, **[4]**, and **[6]** for further details.

7.1. Background

The sample configuration described in these Application Notes explicitly shows a SES combined Home/Edge server configuration. In this case, a single SES server supports both the Home and Edge roles. Multiple SES servers may exist using a separate Home/Edge configuration as warranted by capacity considerations (predominately for the support of SIP phones). In the separate Home/Edge server configuration:

- The SIP signaling relationship with Sprint services exists between the Sprint Border Element and the SES Edge server.
- The Communication Manager SIP signaling group relationship exists between a C-LAN (or equivalent) interface and a specific SES Home server.
- SIP message routing between the Home and Edge servers is performed automatically and transparently. However, administration is required on the Home and Edge servers; refer to **[3]**, **[4]**, and **[6]** for further details.
- Only one SES Edge server exists within a given SIP domain. Multiple Home servers may exist as required.
- All SES administration is performed from a single SES server designated during installation as the master administrator.

The SIP trunking administration is generally the same for the combined and separate Home/Edge configurations. Any specific clarifications will be noted within the individual sections as necessary.

7.2. Host Configuration

1. Launch a web browser, enter http://<IP address of the SES>/admin in the URL, and log in with the appropriate credentials. Click on **Adminstration->SIP Enablement Services**.

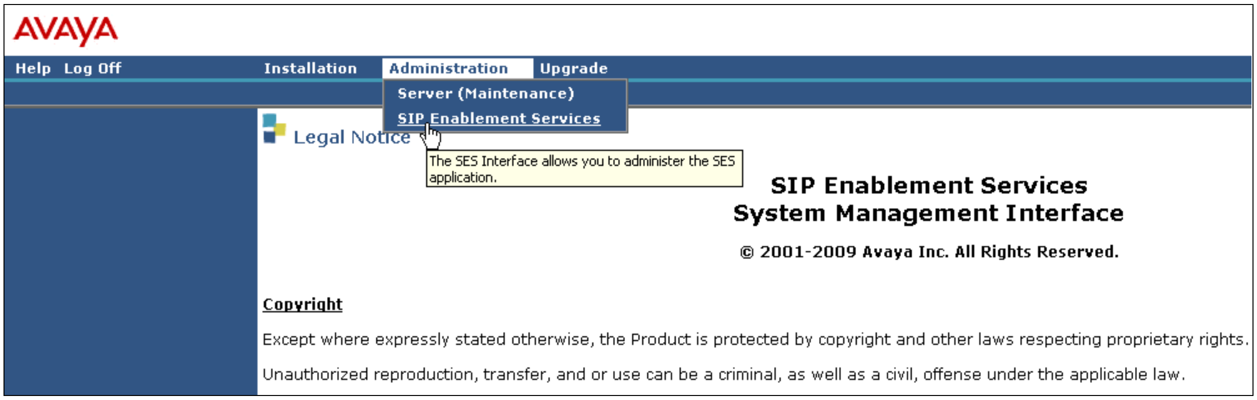

Figure 31: SES Web Interface

2. In the left pane of the SES Administration Interface, expand **Hosts**, and click on "**List**". In the **List Hosts** page, click on "**Edit**".

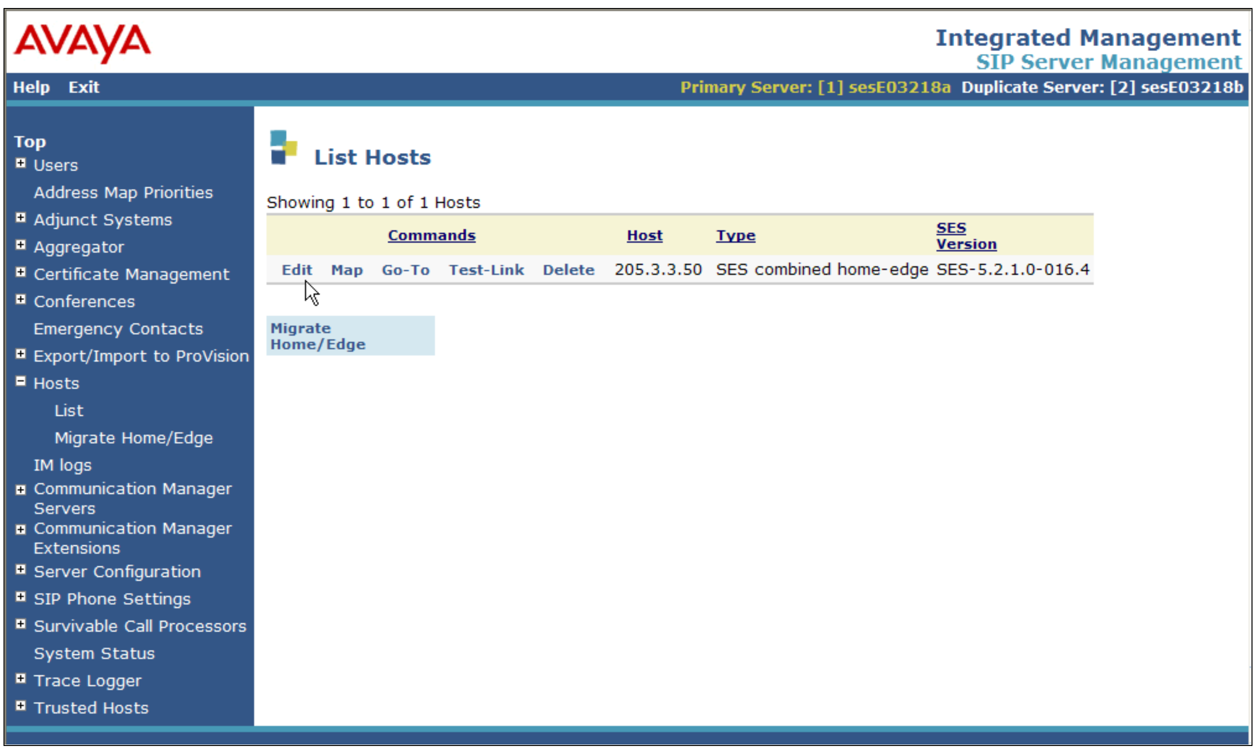

Figure 32: List Hosts Page

3. In the **Edit Host** page, ensure that the following are provisioned:
   - **Host IP address** – Enter the IP address of the SES itself.
   - **Listen Protocols** – The "**UDP**", "**TCP**", and "**TLS**" checkboxes are checked.
   - **Link Protocols** – "**TLS**" is selected.
   - **Outbound Proxy** and **Outbound Direct Domains** fields are blank.

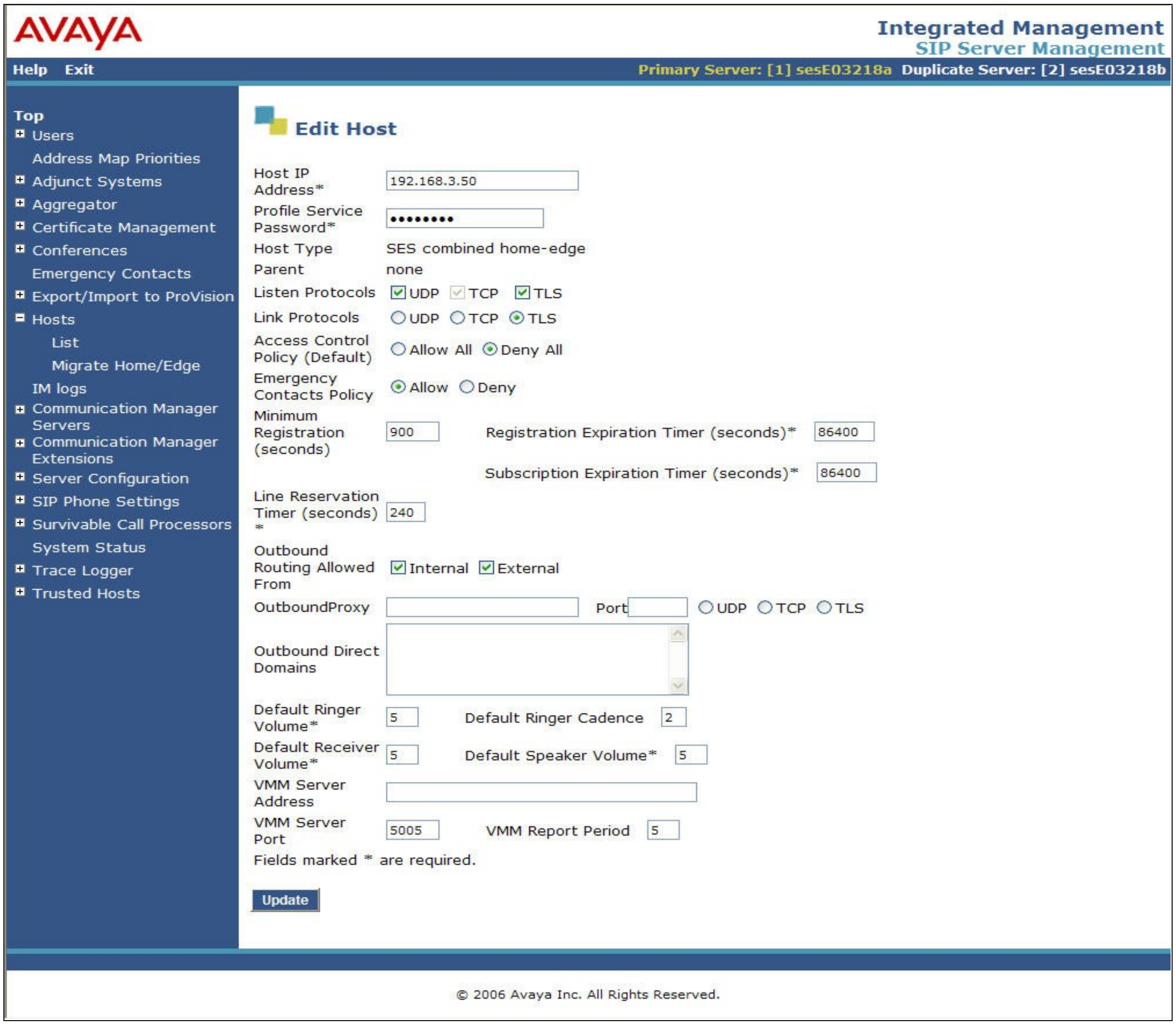

Figure 33: Edit Host Page

Separate Home/Edge Configuration Note:

In the SES separate Home/Edge server configuration, there will be at least two SES hosts listed (Home and Edge servers).

- **SES Edge server** – Configure in the same manner as shown for the combined Home/Edge server above.
- **SES Home servers** – On each SES Home server, set the **Outbound Proxy** to the IP address of the SES Edge server, with **Port** set to "**5061**" and "**TLS**" selected.

7.3. Interfaces to Avaya Aura® Communication Manager

1. In the left pane of the SES Administration Interface, expand **Communication Manager Servers**, and click on "**Add**". In the **Add Communication Manager Server Interface** page, provision the following:
   - **Communication Manager Server Interface Name** – Enter a descriptive name.
   - **Host** – Select the IP address of the SES combined Home/Edge server.
   - **SIP Trunk Link Type** – Select "**TLS**".
   - **SIP Trunk IP Address** – Enter the IP address of the C-LAN board listed in **Section 5.3**, **Step 5**.

   Scroll down to the bottom of the page and click on "**Add**" (not shown). Click on "**OK**" and then "**Continue**" in the subsequent confirmation pages (not shown).

   Separate Home/Edge Configuration Note:
   For SES separate Home/Edge server configurations, select the IP address of an SES Home server for **Host**.

Figure 34: Add Communication Manager Server Interface Page for C-LAN

2. Repeat **Step 1** if additional interfaces to Communication Manager need to be configured.

7.4. Call Routing

7.4.1. Background

SES functions as a SIP proxy for the SIP trunking with Sprint IP Toll Free service. SES examines the Request-URI of a received SIP INVITE message (from the Sprint IP Toll Free service for inbound calls), modifies the Request-URI and certain SIP headers, and then forwards the message to the appropriate destination.

For inbound calls from the Sprint IP Toll Free service, the Request-URI *domain* part contains the IP address of the SES server. Therefore, one or more SES address maps are required to match the *user* part of such inbound calls, and to modify the *domain* part in order to properly route the calls to Communication Manager.

7.4.2. Inbound Calls from Sprint IP Toll Free Service

SES address maps are used to route inbound calls from the PSTN to Communication Manager.

1. In the left pane of the SES Administration Interface, expand **Communication Manager Servers**, and click on "**List**". In the **List Communication Manager Servers** page, click on "**Map**" in the row corresponding to the Communication Manager Server Interface administered in **Section 7.3**, **Step 1**. The Communication Manager Server Map to be added will match inbound PSTN calls with a certain called party number / number range and route those calls to Communication Manager.

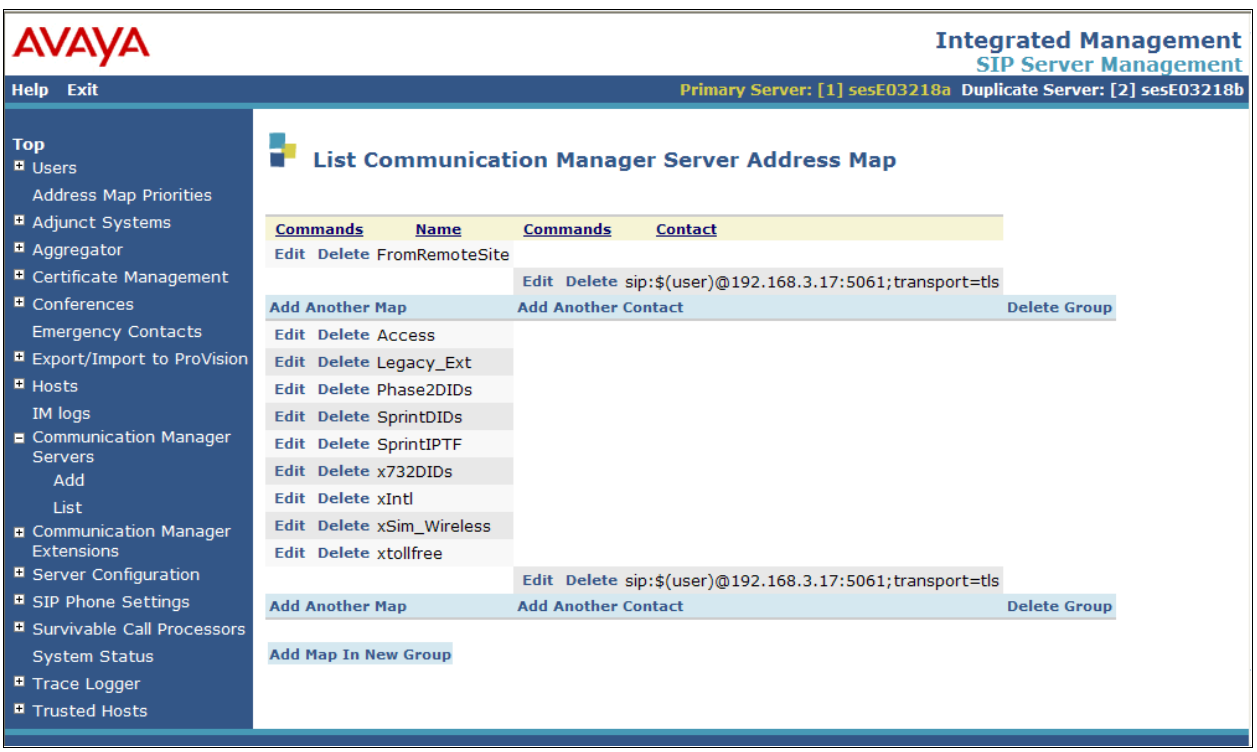

Figure 35: List Communication Manager Server Address Map Page

2. In the **List Communication Manager Server Address Map** page **(**not shown**)**, click on "**Add Map in New Group**".

3. In the **Add Communication Manager Server Address Map** page, provision as follows:
   - **Name** – Enter any descriptive name.
   - **Pattern** – Enter a Linux regular expression that matches the number in the user part of the Request-URI, i.e., the called party number, of inbound SIP INVITE messages for PSTN calls from the Sprint IP Toll Free service. In this sample configuration, for inbound calls from the PSTN, the Sprint IP Toll Free service sends a 10-digit IPTF number beginning with "**85555118**" in the user part of the Request-URI to reach an agent or local extensions in the Avaya site. Thus, the pattern "**^sip:(85555118)[0-9]{2}**" shown below, matches SIP INVITE messages with a Request-URI that may begin with "**sip:85555118**" followed by two digits that can be any number from **0** thru **9**.
     NOTE: In production the customer may only use one toll free number and in that case only one entry for that toll free number would be necessary. A map would not be necessary.
   - Click on "**Add**"
   - Click on "**Continue**" in the subsequent confirmation page (not shown).

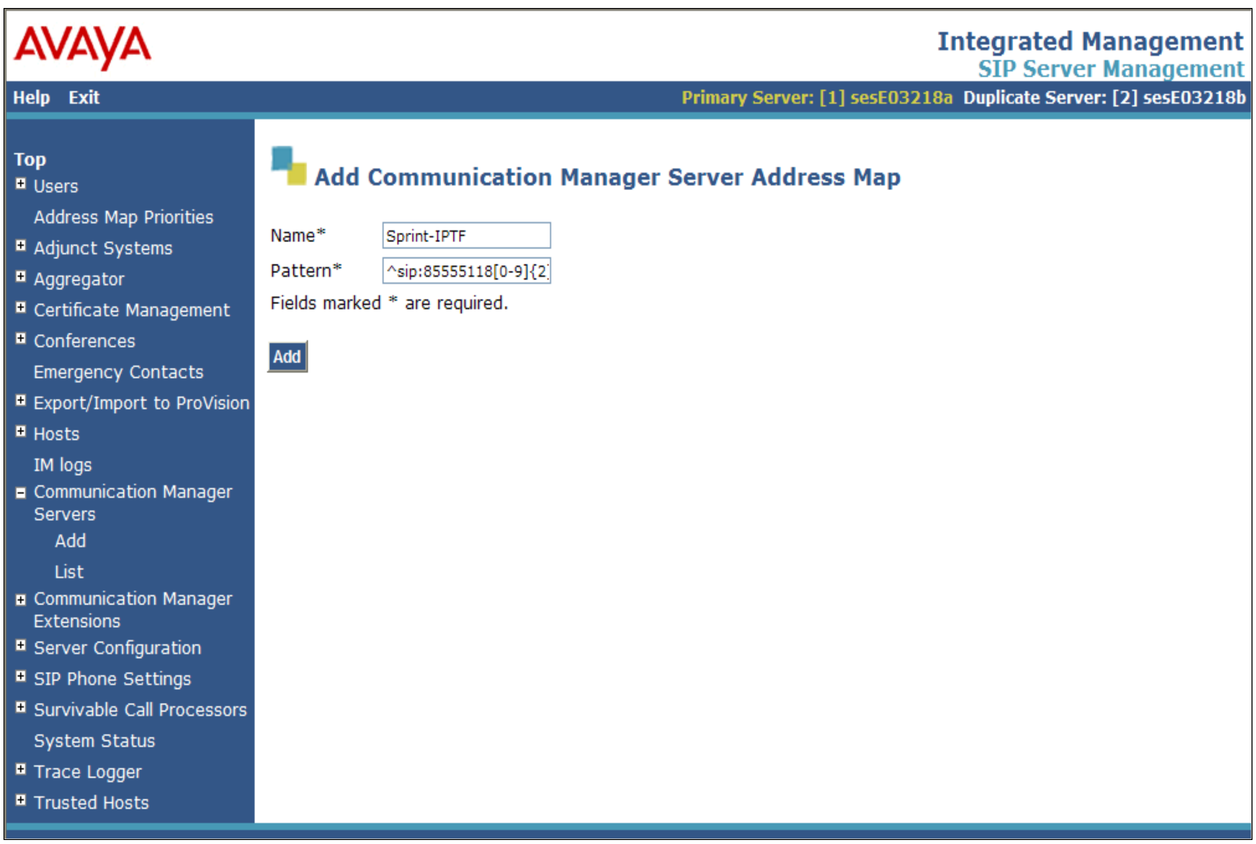

Figure 36: Add Communication Manager Server Address Map Page – Match 9-Digit DNIS

4. A Contact is automatically created after creating the first Communication Manager Server Address Map. The **Contact** specifies that the SIP messages matched by the Communication Manager Server Address Map(s) administered in **Step 3** are to be routed to the IP address of the Communication Manager Server Interface administered in **Section 7.3**, **Step 1**. The **"$(user)"** string in the Contact is substituted with the user part of the original inbound Request-URI.

- Click on "**Add Another Map**"to configure additional address maps to route the calls to Communication Manager or other SIP elements.

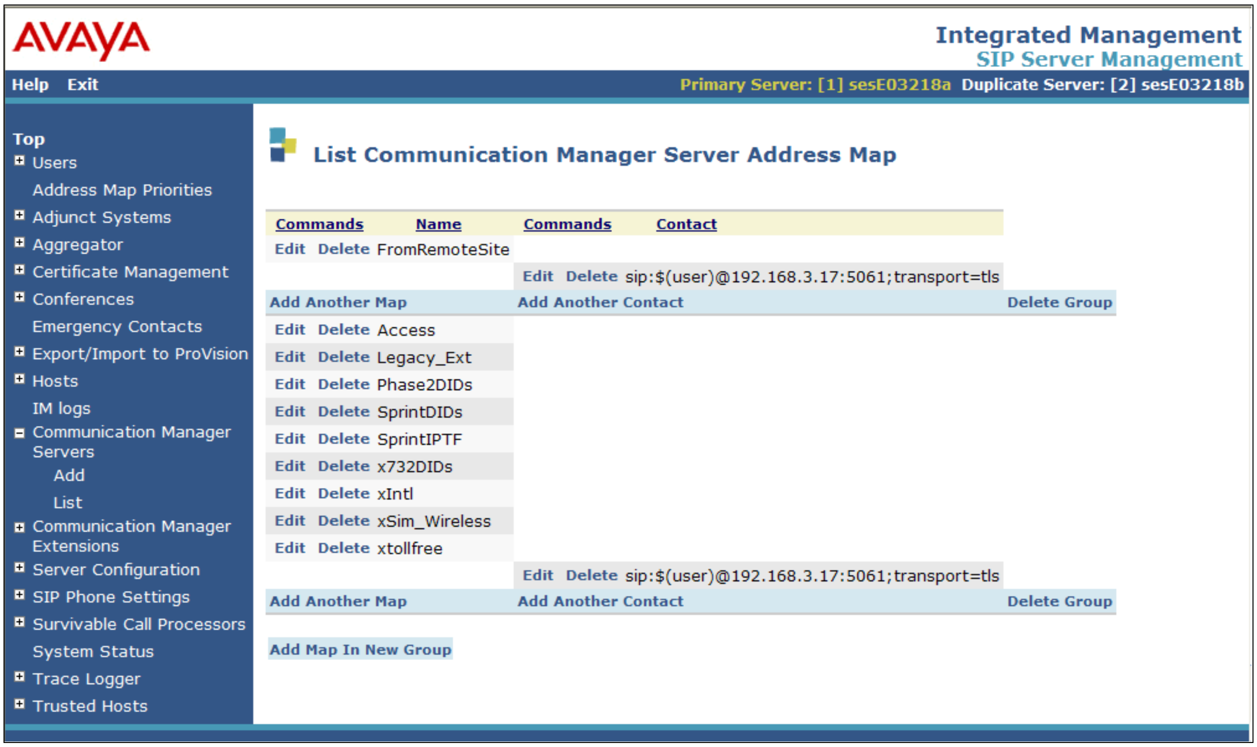

Figure 37: List Communication Manager Server Address Map Page – C-LAN

7.5. Trusted Host

The Sprint IP Toll Free service Session Border Controller IP address must be added as a trusted host entry in SES. SES will not attempt to authenticate incoming requests from trusted hosts.

In the left pane of the SES Administration Interface, expand **Trusted Hosts**, and click on "**Add**". In the **Add Trusted Host** page, provision the following:

- **IP Address** – Enter the IP address of the Sprint IP Toll Free service Session Border Controller.
- **Host** – Select the IP address of the SES combined Home/Edge server.
- **Comment** – Enter a description of the trusted host.
- Click on "**Add**".
- Click on "**Continue**" in the subsequent confirmation page (not shown).
- Repeat the above administration steps for any other Sprint IP Toll Free service Border Elements provided.

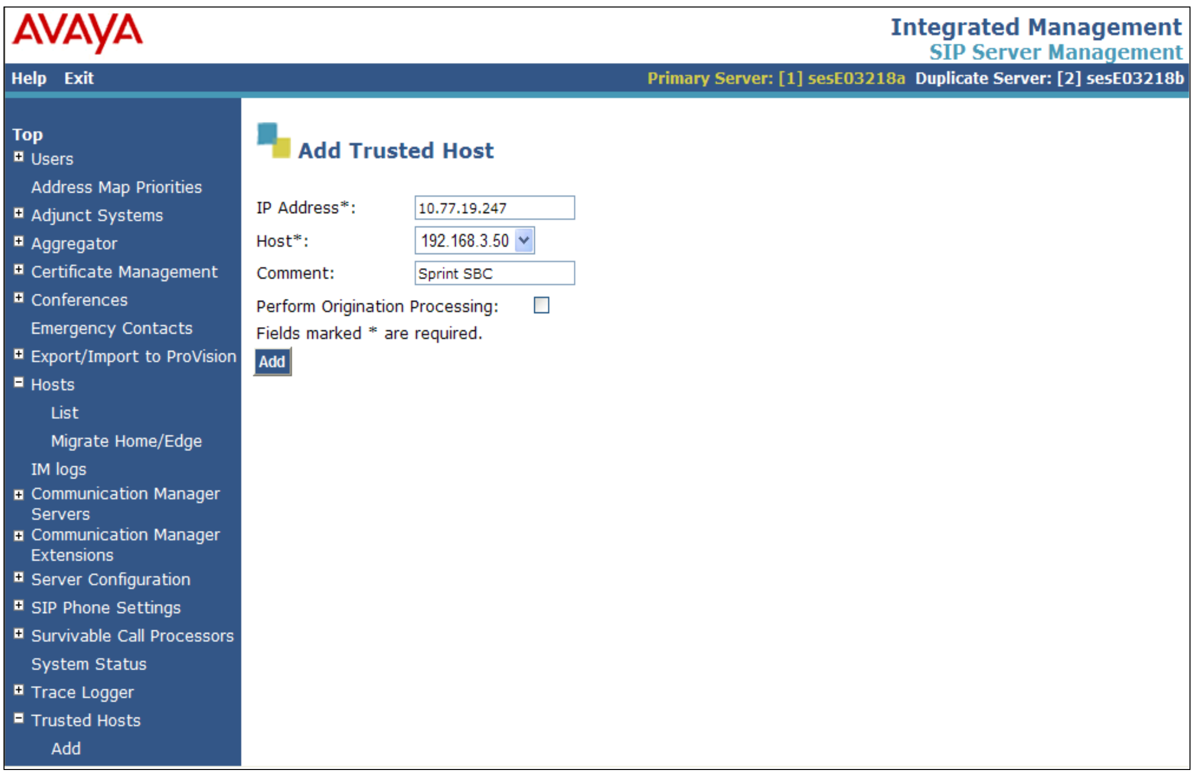

Figure 38: Add Trusted Host Page

Separate Home/Edge Configuration Note:

In the SES separate Home/Edge server configuration, there will be at least two SES hosts listed (Home and Edge servers).

- SES Edge server – Configure the trusted host relationship on the SES Edge server.
- SES Home servers – Configure the trusted host relationship on each SES Home server.

8. Verification Steps

8.1. Verification Tests

The following steps may be used to verify the configuration:

- Place an inbound call, and verify that a two-way talkpath exists, and that the calls remain stable for several minutes and disconnect properly.
- Place an inbound call to an agent or a phone and verify that the call is queued or goes to coverage if it is not answered.

8.2. Troubleshooting Tools

The Communication Manager **list trace vector**, **list trace vdn**, **list trace tac**, and/or **status trunk** ***trunk-group-no*** commands are helpful diagnostic tools to verify correct operation and to troubleshoot problems. MST (Message Sequence Trace) diagnostic traces (performed by Avaya Support) can be helpful in understanding the specific interoperability issues.

The **traceSES** function within the SES may be used to capture SIP traces between SES and the Sprint IP Toll Free service. In addition, if port monitoring is available, a SIP protocol analyzer such as Wireshark (a.k.a. Ethereal) can be used to capture SIP traces at the various interfaces. SIP traces can be instrumental in understanding SIP protocol issues resulting from configuration problems. Note that the SIP messaging between Communication Manager and SES uses TLS encryption and cannot be viewed using Wireshark.

9. Conclusion

These Application Notes described the steps for configuring Avaya Aura® Communication Manager and Avaya Aura® SIP Enablement Services SIP trunking with the Sprint IP Toll Free service. The Sprint IP Toll Free service allows enterprises to receive inbound Toll Free calls from the PSTN. Sprint IP Toll Free Service passed this compliance test.

The sample configuration shown in these Application Notes is representative of a basic enterprise customer configuration and is intended to provide configuration guidance to supplement other Avaya product documentation. It is based upon formal interoperability compliance testing as part of the Avaya DevConnect Service Provider program. Please refer to **Section 2.3** for any observations or workarounds relating to the testing covered by these Application Notes.

10. References

The Avaya product documentation is available at http://support.avaya.com unless otherwise noted.

[1] *Administering Avaya Aura® Communication Manager*, Issue 5.0, Release 5.2, May 2009, Document Number 03-300509

[2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Issue 7, Release 5.2, May 2009, Document Number 555-245-205

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

[4] *Installing, Administering, Maintaining, and Troubleshooting Avaya Aura® SIP Enablement Services*, Issue 7.0, May 2009, Document Number 03-600768

[5] *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide Release 2.5*, Issue 5, November 2009, Document Number 16-601944

[6] *Avaya Aura® SIP Enablement Services (SES) Implementation Guide*, Issue 6, May 2009, Document Number 16-300140

[7] *Avaya Aura® Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference*, Release 5.2, April 2009, Document Number 07-600780

[8] *Avaya Aura® Call Center 5.2 Automatic Call Distribution Reference*, Release 5.2, April 2009, Document Number 07-602568

[9] *Modular Messaging Multi-Site Guide Release 5.2*, November 2009

[10] *Modular Messaging for Microsoft Exchange Release 5.2 Installation and Upgrades*, Issue 1.0, November 2009

Sprint IP Toll Free Service Description:

[11] *Sprint IP Toll Free Service Description*
http://www.sprint.com

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.