

Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the AGN Networks On Demand SIP Service and an Avaya IP Telephony Solution – Issue 1.0

Abstract

These Application Notes describe the steps to configure trunking using the Session Initiation Protocol (SIP) between the AGN Networks On Demand SIP service and an Avaya IP telephony solution. The Avaya solution consists of Avaya SIP Enablement Services, Avaya Communication Manager, and various Avaya SIP, H.323, digital and analog endpoints.

Enterprise customers with this Avaya SIP-based solution can connect via dedicated Internet access using AGN Networks as a service provider to complete PSTN calls. This includes outbound local, long distance and international calling, inbound calling to DID numbers from most major US cities and markets, and inbound toll-free calling. This solution allows customers with a converged network to lower PSTN telecommunication costs, to easily obtain local number presence without offices in each geographic area, and to easily manage their network services using web-based tools.

AGN Networks is a member of the Avaya Developer*Connection* Service Provider program. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps for configuring SIP Trunking between the AGN Networks On Demand SIP Services and an Avaya IP telephony solution consisting of Avaya SIP Enablement Services, Avaya Communication Manager and various Avaya telephony endpoints. These endpoints included IP telephones (using SIP and H.323 protocols), traditional analog and digital phones and the Avaya SIP softphone running on a Microsoft Windows PC.

Customers using this Avaya IP telephony solution with AGN Networks On Demand SIP services are able to place and receive PTSN calls via a dedicated broadband Internet connection using the Session Initiation Protocol (SIP). This converged network solution is an alternative to more traditional PTSN trunks such as T1 or ISDN PRI. It allows customers to possibly reduce local and long distance costs, add and delete DID and toll-free numbers in minutes, as well as benefit from capabilities such as having local numbers from numerous area-codes easily terminate at a single location.

SIP is a signaling protocol designed to provide a common framework for session establishment, modification and termination to support multimedia communications. In this application, SIP acts as the signaling protocol between the Avaya equipment and the network service offered by AGN Networks. SIP manages the establishment of voice connections and the transfer of related information such as calling party identity, etc.

The AGN Networks On Demand SIP family of services covered by this solution includes:

- PacketTrunk - outbound calling to local, long distance and international locations
- PacketDID – incoming Direct Inward Dial (DID) service from most major cities in the US
- PacketTollFree – incoming toll free calling

For the remainder of this document the entire family will simply be referred to as “On Demand SIP” service unless there is a need to differentiate among the services.

Figure 1 illustrates a typical Avaya IP telephony solution connected to AGN Networks On Demand SIP services using SIP trunking. This is the configuration used during the Developer*Connection* compliance testing process.

The Avaya IP telephony solution located on the customer site contained:

- Avaya S8300 Media Server installed into an Avaya G350 Media Gateway. The S8300 served as the host processor for Avaya Communication Manager and for Avaya Intuity Audix 770 integrated voice mail.
- Avaya SIP Enablement Services (SES) software operating on an Avaya S8500B server platform.
- Avaya 4600 series IP telephones (configured to use either the SIP or H.323 protocol).
- Avaya 6400 series digital and 6200 series analog telephones.

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Although not included in **Figure 1**, the enterprise site could also have an alternate route to the PSTN using traditional trunks.

Note: Security devices, such as firewall and network address translation (NAT) devices, were not included in the configuration. These Application Notes focus on SIP trunking interoperability. However, it is recommended that enterprise customers deploy security devices in a production environment.

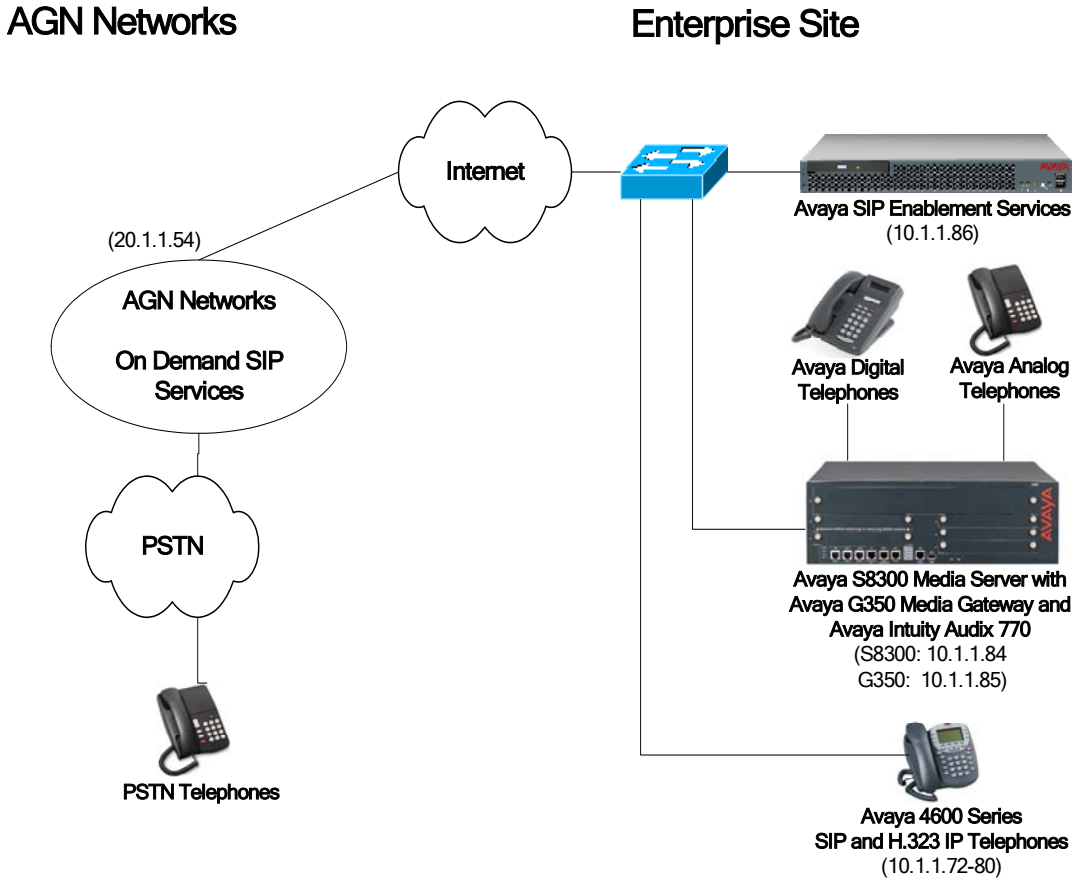


Figure 1: Avaya IP Telephony Network using AGN Networks On Demand SIP Services

1.1. Call Flows

To better understand how calls are routed between the PSTN and the enterprise site shown in **Figure 1** using SIP trunks, two call flows are described in this section.

The first call scenario illustrated in **Figure 2** is a PSTN call to the enterprise site terminating on a typical analog telephone supported by Avaya Communication Manager.

1. A user on the PSTN dials a DID number assigned to an Avaya Communication Manager telephone at the enterprise site. The PSTN routes the call to AGN Networks as the local

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service provider assigned to route the DID number to the assigned customer.

2. Based on the DID number, AGN Networks offers the call to Avaya SES using SIP signaling messages sent via the Internet. Note that the assignment of the DID number and the address of the Avaya SES server was previously established using the AGN Networks web site to administer the On Demand SIP service.

3. Avaya SES routes the call to the Avaya S8300 Media Server running Avaya Communication Manager over a SIP trunk between the elements.

4. Avaya Communication Manager terminates the call to the directly connected analog phone as shown in step 4. The same process occurs for calls to Avaya digital and H.323 IP phones.

   - or –

4a. Inbound calls destined for a SIP extension are routed to Avaya Communication Manager who then transmits the appropriate SIP signaling via Avaya SES to the SIP telephone (as shown by the 4a arrow.)

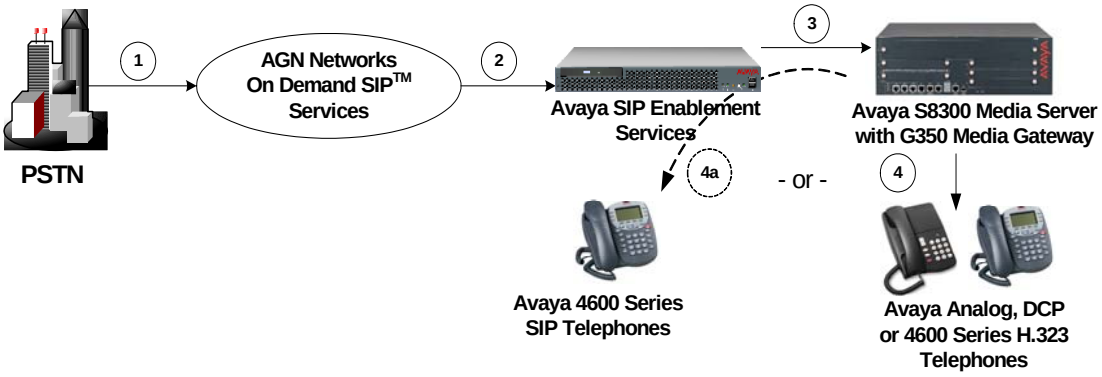


Figure 2: Incoming PSTN Calls to Avaya Communication Manager

The appendix illustrates an example of a SIP INVITE message sent by AGN Networks for an incoming DID call.

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The second call scenario illustrated in **Figure 3** is an outgoing call from an Avaya SIP telephone at the enterprise site to the PSTN via a SIP trunk to AGN Networks.

1. An Avaya H.323, analog or digital telephone served by Avaya Communication Manager originates a call to a user on the PSTN.

   - or-

1a. An Avaya SIP telephone originates a call that is routed via Avaya SES (as shown by the 1a arrow) to Avaya Communication Manager.

2. The call request is handled by Avaya Communication Manager where origination treatment such as class of service restrictions and automatic route selection is performed. Avaya Communication Manager selects the SIP trunk and sends the SIP signaling messages to Avaya SIP Enablement Services.

3. Avaya SIP Enablement Services routes the call to the AGN Networks.

4. AGN Networks completes the call to the PSTN.

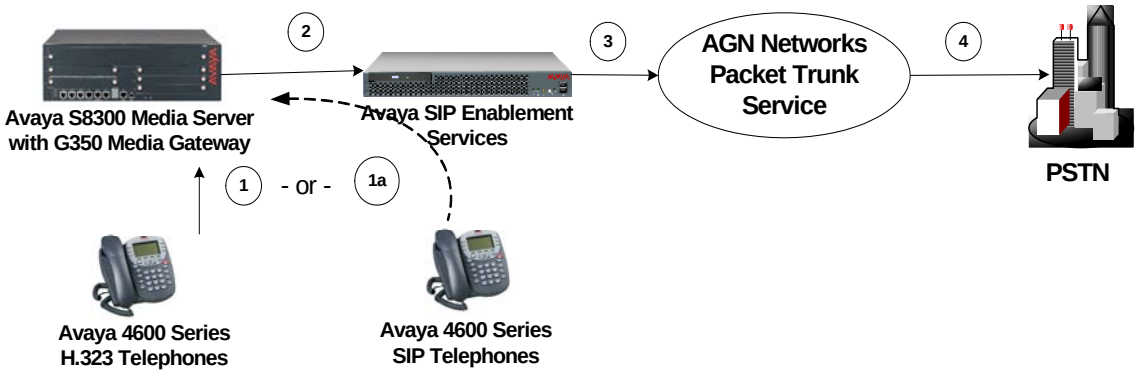


Figure 3: Outgoing Calls from an Avaya Communication Manager to the PSTN

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2. Equipment and Software Validated

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

Hardware Component	Version
Avaya S8300 Media Server with an Avaya G350 Media Gateway and Avaya Intuity Audix 770 (IA770)	Communication Manager 3.0 (R013x.00.0.340.3 update: 00.0.340.3-10673)
Avaya SIP Enablement Services	3.0.0.0-31.0
Avaya 4620 Series SIP Telephones	2.2
Avaya SIP Softphone	Release 2 Build 43
Avaya 4620 Series H.323 IP Telephones	2.3
Avaya 6416 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
AGN Networks SIP Proxy	Nextone MSC v4.0c2-1

3. Configure Avaya Communication Manager

This section describes the steps for configuring a SIP trunk on Avaya Communication Manager. The SIP trunk is established between Avaya Communication Manager and Avaya SIP Enablement Services (SES) server. This trunk will carry the SIP signaling and RTP voice packets sent to AGN Networks.

The SIP trunk also provides the trunking necessary for SIP endpoint devices such as Avaya 4600 SIP telephones and Avaya SIP Softphones to use Avaya Communication Manager in the recommended OPS configuration[1]. It is not necessary to have SIP endpoints in order to use SIP trunking to AGN Networks On Demand SIP services. The steps discussed in Sections 3.2 and 4.2 describing SIP endpoints administration may be omitted if SIP endpoints are not used.

SIP signaling for all incoming calls from On Demand SIP service are received by Avaya SES and routed to Avaya Communication Manager via the SIP trunk. Incoming call treatment, such as performing incoming digit translations, may be performed as necessary. All outgoing calls to the PSTN via On Demand SIP are routed through Avaya Communication Manager in order to use features such as automatic route selection and class of service restrictions. Avaya Communication Manager creates the outbound SIP signaling that is routed via Avaya SES to AGN Networks. Note that Avaya SES acts as a SIP proxy through which all incoming and outgoing SIP messages flow to AGN Networks. There is no direct SIP signaling path between AGN Networks and Avaya Communication Manager or SIP endpoints.

[1] Avaya SIP telephones are configured as off-PBX stations (OPS) on Avaya Communication Manager. Off-PBX stations register with Avaya SES but have calling privileges and features managed by Avaya Communication Manager. Avaya Communication Manager acts as a back-to-back SIP user agent when a SIP phone places or receives a call over a SIP trunk to a service provider.

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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, Operator calls (0), Directory Assistance calls (411) and International calls (011+Country Code) were also supported. Avaya Communication Manager routed all calls using Automatic Route Selection (ARS), except for intra-switch calls. Configuring ARS is beyond the scope of these Application Notes and the reader should refer to [1] for additional information.

Avaya Communication Manager configuration was performed using the System Access Terminal (SAT). The IP network properties of the Avaya S8300 Media Server were configured via its Maintenance web interface using an Internet browser (not shown here).

3.1. SIP Trunk Configuration

Step 1: Confirm Necessary Optional Features

Using the SAT, verify that there exists sufficient SIP Trunks and Off-PBX Telephones capacities by displaying the **System-Parameters Customer-Options** form shown in **Figure 4**. The license file installed on the system controls the maximum permitted. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

On Page 1 of the **System-Parameters Customer-Options** form, verify that the number of OPS stations available is sufficient for the number of SIP telephones to be used.

```
display system-parameters customer-options                    Page   1 of  10
                            OPTIONAL FEATURES
     G3 Version: V13
       Location: 1                             RFA System ID (SID): 1
       Platform: 13                            RFA Module ID (MID): 1

                                                          USED
                                 Platform Maximum Ports: 900   36
                                       Maximum Stations: 40    6
                                 Maximum XMOBILE Stations: 0     0
                     Maximum Off-PBX Telephones - EC500: 50    0
                     Maximum Off-PBX Telephones -   OPS: 50    3
                     Maximum Off-PBX Telephones - SCCAN: 0     0
        (NOTE: You must logoff & login to effect the permission changes.)
```

Figure 4: System-Parameters Customer-Options Form – Page 1

On Page 2, verify that the number of SIP trunks supported by the system is sufficient for the combination of trunks to AGN Networks, SIP endpoints and any other SIP trunks used. Note that each SIP telephone placing calls using AGN Networks requires two SIP trunks for the duration of the call.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 100   6
          Maximum Concurrently Registered IP Stations: 40    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: 100   24

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                            Maximum TN2501 VAL Boards: 0     0
                    Maximum G250/G350/G700 VAL Sources: 1     0
          Maximum TN2602 Boards with 80 VoIP Channels: 0     0
         Maximum TN2602 Boards with 320 VoIP Channels: 0     0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

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

Figure 5: System-Parameters Customer-Options Form – Page 2

Step 2: Assign Node Names

In the **IP Node Names** form, assign the node name and IP address for Avaya SIP Enablement Services at the enterprise site. In this case "SES" and "10.1.1.86" are being used, respectively. The SES node name will be used throughout the other configuration screens of Avaya Communication Manager. In this example "procr" and "10.1.1.84" are the name and IP address assigned to the S8300 Media Server.

```
change node-names ip                                            Page   1 of   1
                                  IP NODE NAMES
    Name              IP Address          Name              IP Address
SES                 10 .1  .1  .86                          .   .   .
default             0  .0  .0  .0                           .   .   .
msgserver           10 .1  .1  .73                          .   .   .
procr               10 .1  .1  .84                          .   .   .
                       .   .   .                            .   .   .
                       .   .   .                            .   .   .
( 4  of 4   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

Figure 6: IP Nodes Names Form

Step 3: Define IP Network Region

The **IP Network Region** form defines the parameters associated with the SIP trunk group serving the Avaya SES proxy (used to reach AGN Networks and any optional SIP endpoints). In the **IP Network Region** form:

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- The **Authoritative Domain** field is configured to match the domain name configured on the Avaya SES. In this configuration, the domain name is *devcon.com*. This field is required for endpoints to call the public network.
- By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between SIP endpoints without using media resources in the Avaya G350 Media Gateway.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within IP network region 1. In this configuration, this codec set will apply to calls with AGN Networks as well as any IP phone (H.323 or SIP) within the enterprise.

  In this case, the SIP trunk is assigned to the same IP network region as the G350 Media Gateway. If multiple network regions are used, Page 3 of each **IP Network Region** form must be used to specify the codec set for inter-region communications.

```
change ip-network-region 1                                    Page   1 of  19
                             IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: devcon.com
    Name:
                               Intra-region IP-IP Direct Audio: yes
MEDIA PARAMETERS               Inter-region IP-IP Direct Audio: yes
      Codec Set: 1                       IP Audio Hairpinning? y
   UDP Port Min: 2048
   UDP Port Max: 3028                 RTCP Reporting Enabled? y
DIFFSERV/TOS PARAMETERS          RTCP MONITOR SERVER PARAMETERS
 Call Control PHB Value: 46       Use Default Server Parameters? y
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6      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 7: IP Network Region Form

Step 4: Define IP Codecs

Open the **IP Codec Set** form using the ip-codec value specified in the **IP Network Region** form (**Figure 7**) and enter the audio codec type to be used for calls routed over the SIP trunk. The recommended settings of the **IP Codec Set** form are shown in **Figure 8.** Note that the **IP Codec Set** form may include multiple codecs listed in priority order to allow the codec for the call to be negotiated during call establishment. For AGN Networks, only G.711MU, G.711A or G.729A can be included in this list.

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

                          IP Codec Set

    Codec Set: 1

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

Figure 8: IP Codec Set Form

Step 5: Configure the Signaling Group
Configure the **Signaling Group** form shown in **Figure 9** as follows:

- Set the **Group Type** field to *sip*.
- The **Transport Method** field will default to *tls* (Transport Layer Security). TLS is the only link protocol that is supported for SIP trunking with Avaya SIP Enablement Services.
- Specify the Avaya S8300 Media Server (node name "procr") and the Avaya SIP Enablement Services Server (node name "SES") as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** fields, respectively. These field values are taken from the **IP Node Names** form shown in **Figure 6**. For other media server platforms, the near (local) end of the SIP signaling group may be a C-LAN board instead of the media server.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- Enter the IP Network Region value assigned in the ip-network-region form (**Figure 7**). Note that if the **Far-end Network Region** field is different from the near-end network region, the preferred codec will be selected from the IP codec set assigned for the inter-region connectivity for the pair of network regions. In this case, the same ip network region (Network Region 1) was used for local and PSTN calls; however, different network regions can be used in the field.
- Enter the domain name of Avaya SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is *devcon.com*. This domain is specified in the Uniform Resource Identifier (URI) of the "SIP To Address" in the INVITE message. Mis-configuring this field may prevent calls from being successfully established to other SIP endpoints or to the PSTN.
- If calls to/from SIP endpoints are to be shuffled, then the **Direct IP-IP Audio Connections** field must be set to 'y'.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833.
- The default values for the other fields may be used.

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


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```
                               SIGNALING GROUP

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


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

                                           Bypass If IP Threshold Exceeded? n

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

Figure 9: Signaling Group Form

Step 6: Configure the Trunk Group

Configure the **Trunk Group** form as shown in **Figure 10** using the "add trunk-group" command. In this case the trunk group number chosen is 1. On Page 1 of this form:

- Set the **Group Type** field to *sip*.
- Choose a mnemonic **Group Name**.
- Specify an available trunk access code (**TAC**).
- Set the **Service Type** field to *tie*.
- Specify the signaling group associated with this trunk group in the **Signaling Group** field as previously specified in **Figure 9**.
- Specify the **Number of Members** supported by this SIP trunk group.

  Note that one trunk member is required for each call between a non-SIP endpoint and AGN Networks. Calls involving a SIP endpoint and AGN Networks will use two trunk members for the duration of the call.

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```
add trunk-group 1                                          Page   1 of  19
                              TRUNK GROUP

Group Number: 1                    Group Type: sip           CDR Reports: y
  Group Name: To SES                        COR: 1        TN: 1        TAC: 101
   Direction: two-way        Outgoing Display? n
 Dial Access? n                Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                                      Signaling Group: 1
                                                    Number of Members: 24
TRUNK PARAMETERS

     Unicode Name? y

                                          Redirect On OPTIM Failure: 5000

            SCCAN? n                                Digital Loss Group: 18
```

Figure 10: Trunk Group Form – Page 1

On Page 2 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 1                                          Page   2 of  19
TRUNK FEATURES
          ACA Assignment? n           Measured: none
                                                      Maintenance Tests? y


                     Numbering Format: public

                                               Replace Unavailable Numbers? n
```

Figure 11: Trunk Group Form – Page 2

Step 7: Configure Calling Party Number Information

Configure the **Numbering Public/Unknown Format** form to send the full calling party number to the far-end.

In this case, all stations with a 5-digit extension beginning with 7 should send the calling party number 732-20x-xxxx when an outbound call uses SIP trunk group #1. This calling party number will be sent to the far-end in the SIP "From" header.

Figure 12 shows the use of the "change public-unknown numbering" command to implement this rule.

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

 5  7      1         73220           10
```

Figure 12: Numbering Public/Unknown Format Form

Step 8: Configure Incoming Digit Translation
This step performs the steps necessary to map incoming DID calls to the proper extension(s).

The incoming digits (specified using the AGN Networks **Desired DNIS** value shown in **Figure 45**) are manipulated as necessary to route calls to the proper extension on Avaya Communication Manager. Note that this step cannot be completed until the DID numbers and routing strategy defined in Sections 4.1 and 5 is known. Return to this step after the Section 5 work is completed if necessary.

In the examples used in these Application Notes, the incoming DID numbers assigned by AGN Networks do not have a direct correlation to the internal extensions assigned within Avaya Communication Manager. Thus all incoming called number digits are deleted and replaced by the assigned extension number.

To create a fully mapped extension number as shown in **Figure 13**:

- Open the **Incoming Call Handling Treatment** form for the SIP trunk group.
- For each extension, from the AGN Networks administration shown in **Figure 45**, enter the length of the incoming **Desired DNIS** value into the **Called Len** and **Del** fields, and the entire **Desired DNIS** value into the **Called Number** field. Enter the desired Avaya Communication Manager extension number into the **Insert** field.

```
change inc-call-handling-trmt trunk-group 1                    Page   1 of   3
                      INCOMING CALL HANDLING TREATMENT
 Service/       Called    Called        Del  Insert
 Feature         Len      Number
 tie              14  90119087650215    14   70000
 tie              14  90119087650216    14   72000
 tie              14  90119087650219    14   72001
```

Figure 13: Incoming Call Handling Treatment – Full Extension Mapping

If the customer's extension numbering aligns with the DID numbers (i.e. the last four DID digits match the extension, it is not necessary to define an entry for each DID number. Those entries might be similar to **Figure 14** where a match against the incoming routing prefix and npa/nnx could be done to route to a correlated extension.

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```
change inc-call-handling-trmt trunk-group 1                    Page   1 of   3
                      INCOMING CALL HANDLING TREATMENT
 Service/       Called    Called        Del  Insert
 Feature         Len      Number
 tie              14    9011908765      10
```

Figure 14: Incoming Call Handling Treatment – Simple Extension Mapping

Step 9: Save Avaya Communication Manager Changes
Enter "save translation" to make the changes permanent.

3.2. SIP Endpoint Configuration

This section describes the administration of SIP telephones and requires the preceding SIP Trunk configuration to have been completed. The use of SIP telephones is optional; they are not required to use the AGN Networks On Demand SIP service.

Step 1: **Assign a Station**
The first step in adding an off-PBX station (OPS) for Avaya SIP telephones registered with Avaya SIP Enablement Services is to assign a station as shown in **Figure 15**.

Using the "add station" command from the SAT:

- Leave the station **Type** to at the default "6408D+" value. (Note this is the Avaya recommended best practice that will prevent alarm warning that occur when 4600 series phone models are entered).
- Enter "X" in the **Port** field to indicate station administration without port hardware.
- Enter a **Name** for the station that will be displayed.
- The **Security Code** is left blank for SIP OPS extensions.

The remaining fields are configured per normal station administration that is beyond the scope of these Application Notes. Note that the Class of Restrictions (**COR**) and Class of Service (**COS**) will govern the features and call restrictions that apply to this station. This station is also configured with voicemail coverage via coverage path 1 using the Intuity Audix 770, an option integrated into the Avaya S8300 Media Server. For additional information on configuring Intuity Audix 770, refer to [4].

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```
add station 70000                                          Page   1 of   4
                                   STATION

Extension: 70000                         Lock Messages? n          BCC: 0
     Type: 6408D+                          Security Code:             TN: 1
     Port: X                           Coverage Path 1: 1          COR: 1
     Name: SIP70000                    Coverage Path 2:            COS: 1
                                       Hunt-to Station:
STATION OPTIONS
              Loss Group: 2            Personalized Ringing Pattern: 1
            Data Module? n                        Message Lamp Ext: 70000
          Speakerphone: 2-way                  Mute Button Enabled? y
       Display Language: English
                                              Media Complex Ext:
                                                   IP SoftPhone? n
```

Figure 15: Station Administration – Page 1

On Page 2 of the **Station** form,

- Set the **Restrict Last Appearance** value to 'n' on phones that have 3 or fewer call appearances to maintain proper SIP conference and transfer operation.

  Setting the **Restrict Last Appearance** value to 'y' reserves the last call appearance for outbound calls. Certain SIP conference and transfer features will not function properly if a third appearance is not available for incoming calls.

- 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 70000                                          Page   2 of   4
                                   STATION
FEATURE OPTIONS
          LWC Reception: spe             Auto Select Any Idle Appearance? n
         LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                             Auto Answer: none
            CDR Privacy? n                          Data Restriction? n
   Redirect Notification? y                 Idle Appearance Preference? n
 Per Button Ring Control? n               Bridged Idle Line Preference? n
   Bridged Call Alerting? n                   Restrict Last Appearance? n
  Active Station Ringing: single          Conf/Trans on Primary Appearance? n

       H.320 Conversion? n          Per Station CPN - Send Calling Number? y
      Service Link Mode: as-needed
        Multimedia Mode: enhanced
    MWI Served User Type: qsig-mwi             Display Client Redirection? n
                                             Select Last Used Appearance? n
                                               Coverage After Forwarding? s

 Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
 Emergency Location Ext: 70000     Always Use? n      IP Audio Hairpinning? y
```

Figure 16: Station Administration – Page 2

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On Page 3 of the **Station** form, configure the appropriate number of call appearances for the SIP telephone as shown in **Figure 17**.

```
add station 70000                                        Page   3 of   4
                                    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 17: Station Adminstration – Page 3

A similar number of call appearances should be configured on the SIP Telephone which is beyond the scope of these Application Notes. The parameters to administer call appearances (and many other settings) further described in Reference [6].

Step 2: **Configure Off-PBX Station Mapping**

The second step of configuring an off-PBX station is to configure the **Off-PBX Telephone** form so that calls destined for a SIP telephone at the enterprise site are routed to Avaya SIP Enablement Services, which will then forward the call to the SIP telephone.

On the **Off-PBX-Telephone Station-Mapping** form shown in **Figure 18**:

- Specify the **Station Extension** of the SIP endpoint.
- Set the **Application** field to *OPS*.
- Set the **Phone Number** field to the digits to be sent over the SIP trunk. In this case, the SIP telephone extensions configured on Avaya SIP Enablement Services also match the extensions of the corresponding AWOH stations on Avaya Communication Manager. However, this is not a requirement.
- Set the **Trunk Selection** field to '1', which is the number assigned to the SIP trunk group used to route the call to the SIP station. This trunk group number was previous defined in **Figure 10**.
- Set the **Configuration Set** to the value '1', which in these Application Notes uses the default values of the Configuration Set form.

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```
change off-pbx-telephone station-mapping 70000                  Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station     Application   Dial   Phone Number      Trunk       Configuration
 Extension                 Prefix                   Selection   Set
 70000         OPS              - 70000              1          1
                                -
```

Figure 18: 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 70000                  Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged
 Extension     Limit      Mode         Allowed     Calls
 70000         3          both         all         both
```

Figure 19: Stations with Off-PBX Telephone Integration – Page 2

Step 3: Repeat for each SIP Phone
Repeat Steps 1 and 2 for each SIP phone to be added.

Step 4: Save Avaya Communication Manager Changes

Enter "save translation" to make the changes permanent.

4. Configure Avaya SIP Enablement Services

This section covers the administration of Avaya SIP Enablement Services (SES). Avaya SIP Enablement Services is configured via an Internet browser using the Administration web interface. It is assumed that Avaya SIP Enablement Services software and the license file have already been installed on Avaya SIP Enablement Services. During the software installation, the ccsInstaller script is run on the Linux shell of the server to specify the IP network properties of the server along with other parameters. For additional information on these installation tasks, refer to [3].

This section is divided into two parts: Section 4.1 provides the steps necessary to configure SIP trunking to AGN Networks. Section 4.2 provides the steps necessary to complete the administration for optional SIP endpoints (whose configuration was begun on Avaya Communication Manager in Section 3.2).

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4.1. SIP Trunking to AGN Networks

Step 1: Log in to Avaya SIP Enablement Services

Access the SES Administration web interface, by entering http://<ip-addr>/admin as the URL in an Internet browser, where *<ip-addr>* is the IP address of Avaya SIP Enablement Services.

Log in with the appropriate credentials and then select the *Launch Administration Web Interface* link from the main screen as shown in **Figure 20**.

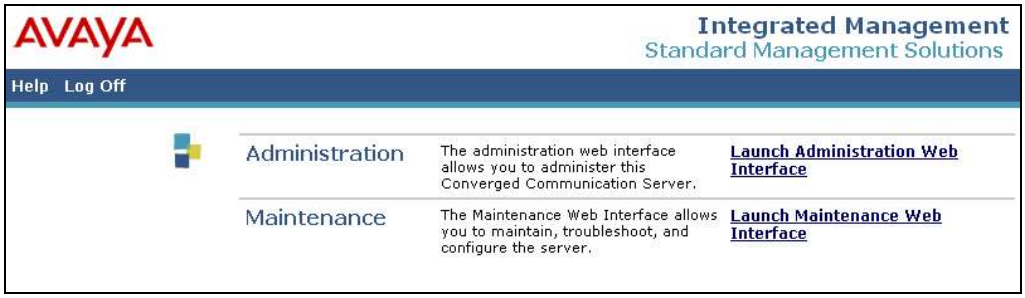


Figure 20 - Avaya SES Main Screen

The SES administration home screen shown in **Figure 21** should be displayed.

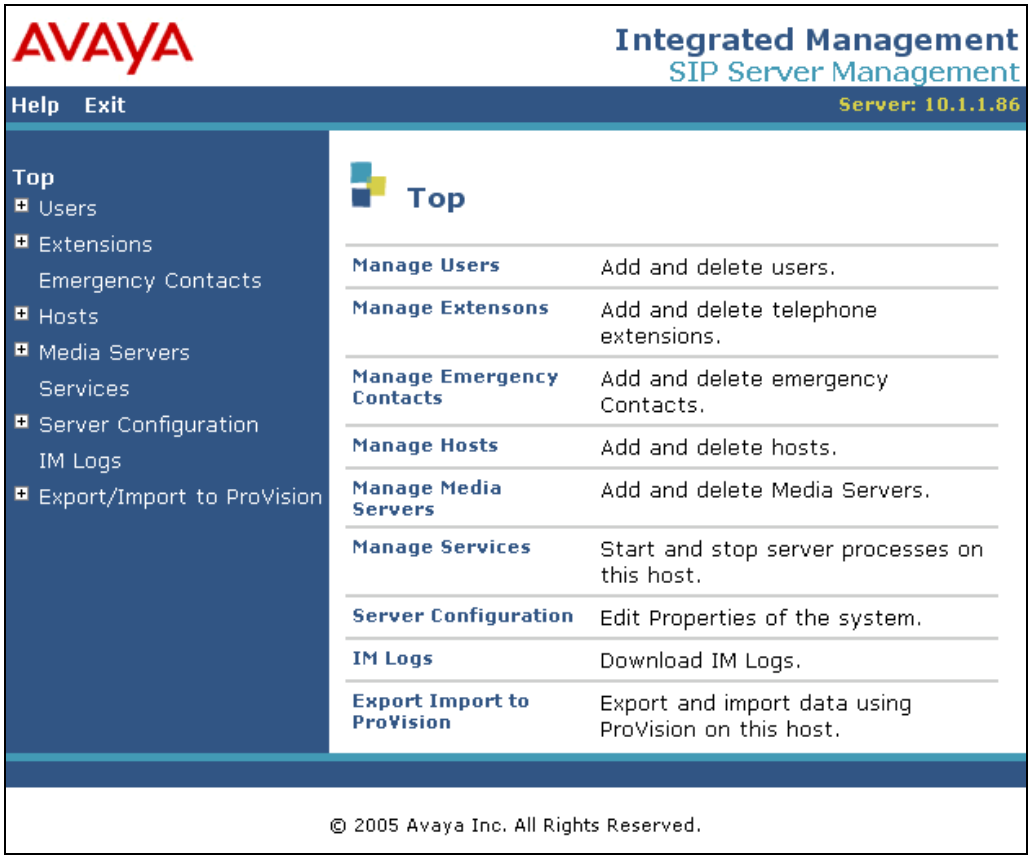


Figure 21: Avaya SES Administration Home Page

Step 2: Define System Properties
From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. This screen displays the CCS version and the network properties entered via the **ccsInstaller** script during the installation process.

In the **System Properties** screen,

- Enter the **SIP Domain** name assigned to Avaya SIP Enablement Services.
- Enter the **License Host** field. This is the host name, the fully qualified domain name, or the IP address of the SIP proxy server that is running the WebLM application and has the associated license file installed. This entry should always be **localhost** unless the WebLM server is not co-resident with this server.
- After configuring the **System Properties** screen, click the **Update** button.

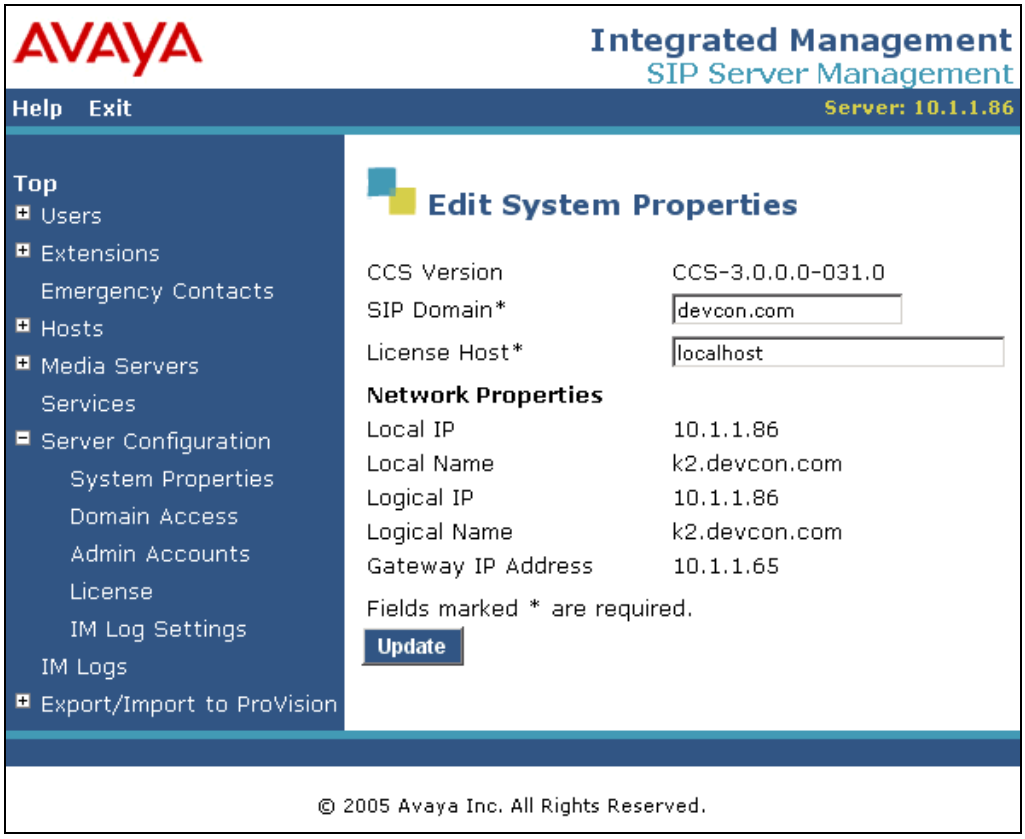


Figure 22: System Properties

Step 3: Enter Avaya SES Host Information
After setting up the domain in the **System Properties** screen, create a host computer entry for Avaya SIP Enablement Services. The following example shows the **Edit Host** screen since the host had already been added to the system.

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The **Edit Host** screen shown in **Figure 23** is accessible by clicking on the **Hosts** link in the left pane and then clicking on the **edit** option under the **Commands** section of the subsequent page that is displayed.

- Enter the **Logical IP** or **Logical Name** (shown in **Figure 22**) of this server in the **Host IP Address** field.
- Enter the **DB Password** that was specified while running the **ccsInstaller** script during the system installation.
- Configure the **Host Type** field. In this example, the host server was configured as a *home/edge* since no additional SES proxy servers exist in the enterprise network.
- The default values for the other fields may be used as shown in **Figure 23**.
- Click the **Update** button.

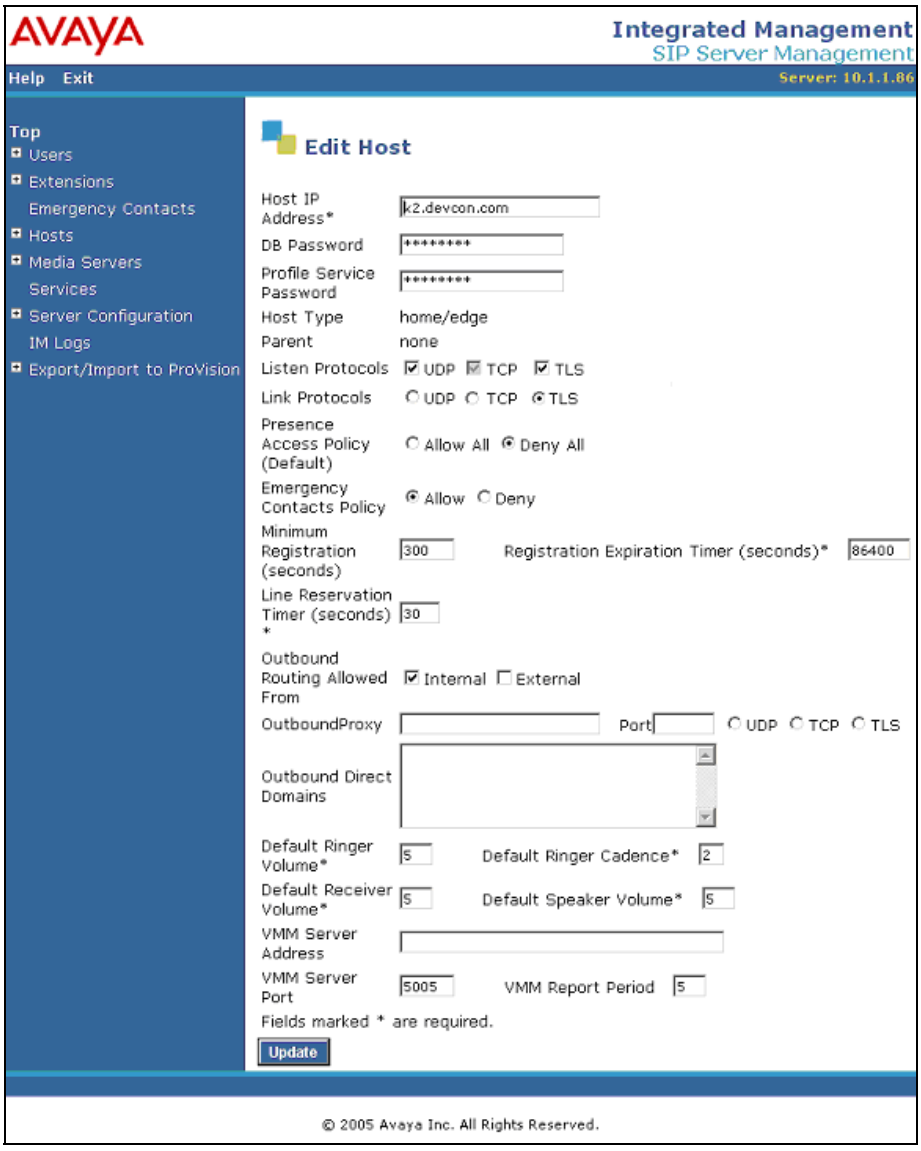


Figure 23: Edit Host

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Step 4: **Add Avaya Communication Manager as Media Server**
Under the **Media Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server in the enterprise site. This will create the Avaya SES side of the SIP trunk previously created in Avaya Communication Manager.

In the **Add Media Server** screen, enter the following information:

- A descriptive name in the **Media Server Interface** field (e.g., S8300).
- Select the home SES server in the **Host** field as specified in **Figure 23**.
- Select *TLS* (Transport Link Security) for the **Link Type**. TLS provides encryption at the transport layer. TLS is the only link protocol that is supported for SIP trunking with Avaya Communication Manager.
- Enter the IP address of the Avaya S8300 Media Server in the **SIP Trunk IP Address** field. (Note: This may be the IP address of a C-LAN board in larger Avaya Communication Manager configurations such as an Avaya S8710 Media Server using an Avaya G650 Media Gateway.)
- After completing the **Add Media Server** screen, click on the **Add** button.

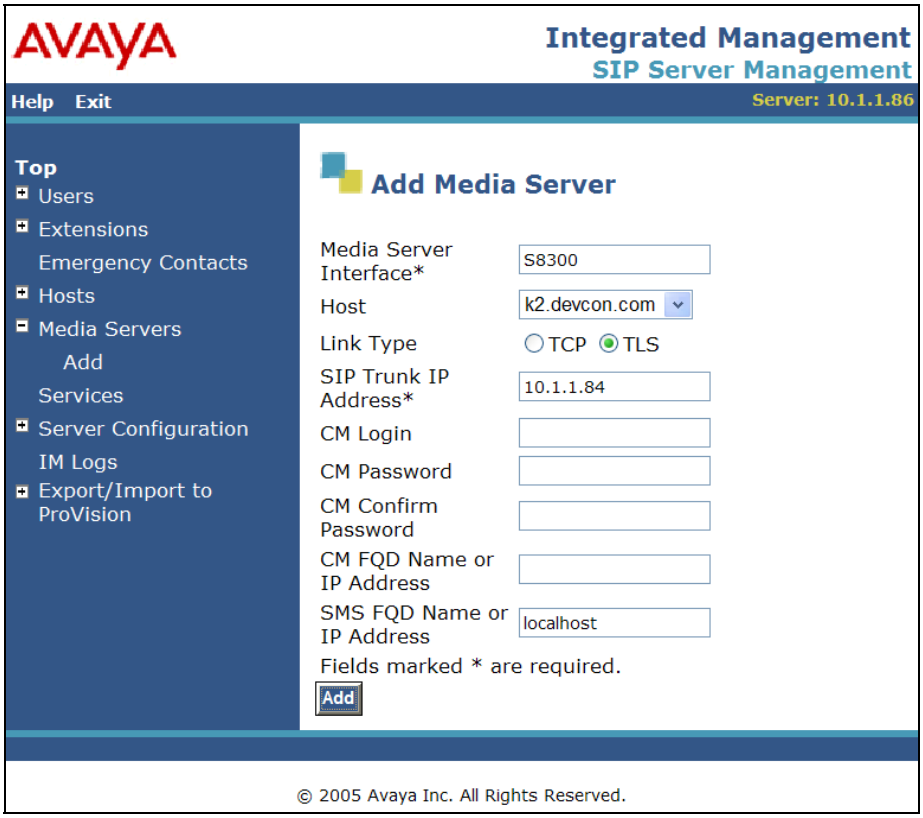


Figure 24: Add Media Server

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Step 5: Specify Address Maps to Media Servers
Incoming calls arriving at Avaya SIP Enablement Services are routed to the appropriate Avaya Communication Manager for termination services. This routing is specified in a Media Server Address Map configured on Avaya SIP Enablement Services.

This routing compares the Uniform Resource Identifier (URI) of an incoming INVITE message to the pattern configured in the Media Server Address Map**,** and if there is a match, the call is routed to the designated Avaya Communication Manager. The URI usually takes the form of *sip:user@domain*, where *domain* can be a domain name or an IP address. Patterns must be specific enough to uniquely route incoming calls to the proper destination if there are multiple Avaya Communication Managers supported by the Avaya SES server.

In these Application Notes, only incoming calls from the PSTN require a media server address map entry. Calls originated by Avaya SIP telephones configured as OPS are automatically routed to the proper Avaya Communication Manager by the assignment of an Avaya Media Server extension to that phone. Address map definitions for SIP endpoints not assigned a media server extension and connections to multiple service providers are beyond the scope of these Application Notes.

For the On Demand SIP service, the *user* portion of the SIP URI will contain the "Desired DNIS" value specified for the incoming direct inward dialed telephone number. The "Desired DNIS" is administered on the AGN Networks customer web site as discussed in Section 5 - **Figure 45**.

An example of a SIP URI in an INVITE message received from AGN Networks would be:

sip:90112144524121;npdi=yes@10.1.1.86.

The user portion containing the specified "Desired DNIS" value in this case is "90112144524121".

To minimize the complexity of the address maps but permit full flexibility in the assignment of any DID number to any media server, the Desired DNIS value was defined to be a 3-digit routing number followed by the full 11 digit North American telephone number. In this case, routing number 901 will be used for DID numbers assigned to the first media server, 902 for the second media server, etc.

Thus the media server address map strategy will be to define pattern matches for the 3-digit routing number in the URI and have the SES forward the messages that match to the appropriate media server.

To configure a **Media Server Address Map**:

- Select **Media Servers** in the left pane of the Administration web interface. This will display the **List Media Servers** screen.

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- Click on the **Map** link associated with the appropriate media server to display the **List Media Server Address Map** screen.
- Click on the **Add Map In New Group** link. The screen shown in **Figure 25** is displayed. The **Host** field displays the name of the media server that this map applies to.
- Enter a descriptive name in the **Name** field
- Enter the regular expression to be used for the pattern matching in the **Pattern** field.

  In this configuration, the pattern specification (without the double quotes) for DID numbers assigned to the 901 group is: `"^sip:9011[0-9]{10}"`. This says for URIs beginning with "sip:9011" followed by any combination of "10 digits" should be routed to the S8300 media server.

  Appendix B provides a detailed description of the syntax for address map patterns.

- Click the **Add** button once the form is completed.

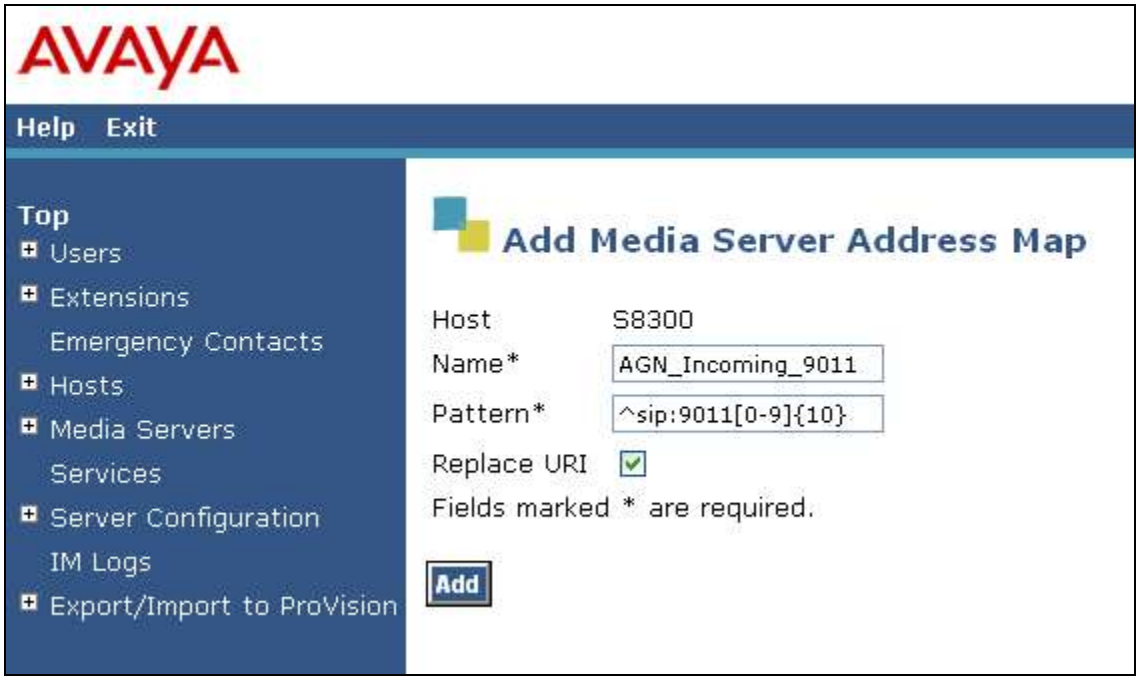


Figure 25: Media Server Address Map

After configuring the media server address map, the **List Media Server Address Map** screen appears as shown in **Figure 26**.

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Figure 26: List Media Server Address Map

Note that after the first **Media Server Address Map** is added, the **Media Server Contact** is created automatically. For the **Media Server Address Map** added in **Figure 25**, the following contact was created:

```
sips:$(user)@10.1.1.84:5061;transport=tls
```

The contact specifies the IP address of the Avaya S8300 Media Server and the transport protocol used to send SIP signaling messages. The user in the original request URI is substituted for `$(user)`.

Step 6: Specify Address Maps to AGN Networks

Outbound PSTN calls are first directed by Avaya Communication Manager routing decisions to the SIP trunk group. These calls are then subject to further routing decisions determined by the Host Address Maps in Avaya SIP Enablement Services. Similar to the inbound media server address maps, these Host Address Maps use pattern matching to direct outbound SIP messages to the proper destination.

In this configuration, the routing rule for the SIP trunk group will be to send all outbound PSTN traffic to the On Demand SIP service. To perform this, two dialing patterns will be created in Avaya SES. The first pattern (without the double quotes) of "^sip:1[0-9]{10}" will match on all sip calls having 1 followed by any 10 digits. The second pattern of "^sip:0" will route any sip call beginning with 0 (regardless of the following digits). Note that additional or more specific pattern matches would be used if necessary to selectively route SIP traffic to different destinations (such as multiple service providers serving different geographic regions). Also note

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that a user dialed access code (such as 9 to place a PTSN call) has been previously deleted prior to seizing the outbound SIP trunk.

The configuration of the host address map for all North America calls is shown in **Figure 27**.

- Access the **Add Host Address Map** screen by selecting the **Hosts** link in the left pane of the Administration web interface and then clicking on the **Map** link associated with the appropriate host (e.g., k2.devcon.com). The **List Host Address Map** screen is displayed.
- From this screen, click the **Add Map In New Group** link to display the **Add Host Address Map** screen shown in **Figure 27**.
- Enter a descriptive name for the map, such as "AGN_Out1Plus10".
- Specify an appropriate pattern for the call type. In this example, the pattern is used for North American calls is "`^sip:1[0-9]{10}`".
- Leave the **Replace URI** checkbox selected.
- Click the **Add** button.



Figure 27: Edit Host Map Entry

Additional Host Address Map patterns are added in the same manner. To complete the example and allow all "0" and "0+any digits" calls directed to the SIP trunk group to be sent to AGN Networks, add another entry using the pattern "^sip:0" to the Host Address Map list. The resulting screen is shown in **Figure 28**.

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Figure 28: Multiple Address Maps

Step 7: Specify the AGN Networks Host Contact Information
The next step is to enter the contact address for the AGN Networks SIP proxy. In this example, the IP address 20.1.1.54 is used. The customer's specific information will be provided by AGN Networks.

To enter the AGN Networks SIP proxy information:

- As described in Step 6, display the **List Host Address Map** screen.
- Click on the **Add Another Contact** link associated with the address map added in **Figure 27** and **Figure 28** to open the **Add Host Contact** screen. In this screen, the **Contact** field specifies the destination for the call and it is entered as:

  ```
  sip:$(user)@20.1.1.54:5060;transport=udp
  ```

  Note: Replace the IP address (20.1.1.54) with the information provided by AGN Networks.

  The user part in the original request URI is inserted in place of the "$(user)" string before the message is sent to AGN Networks.
- Click the **Add** button when completed.

After configuring the host address maps, the **List Host Address Map** screen will appear as shown in **Figure 29**.

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Figure 29: List Host Address Map

Step 8: Save the Changes

After making changes within Avaya SES, it is necessary to commit the database changes using the **Update** link that appears when changes are pending. Perform this step by clicking on the **Update** link found in the bottom of the blue navigation bar on the left side of any of the SES Administration screens as shown in **Figure 30**.

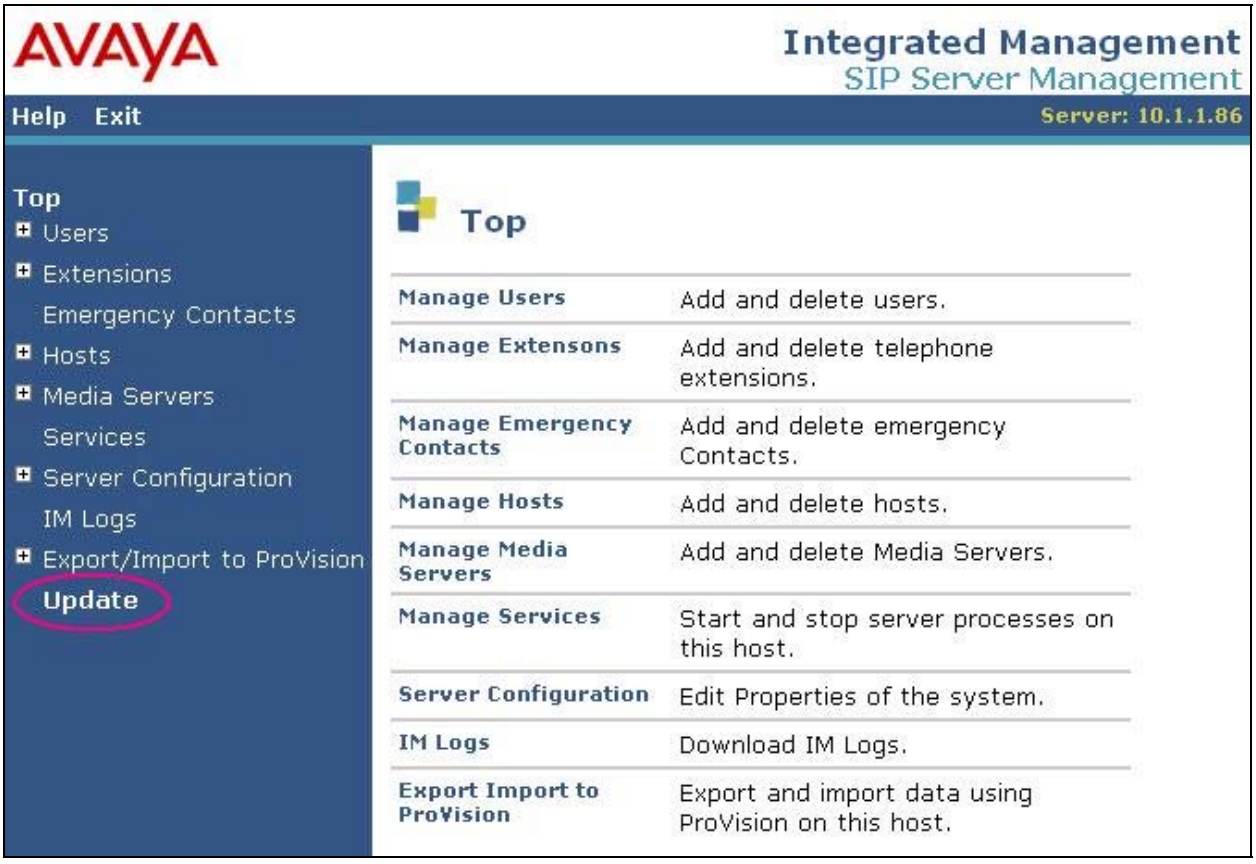


Figure 30: Update Following SES Administrative Changes

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Step 9: Specify AGN Networks as a Trusted Host

The final step to complete the SIP trunk administration on Avaya SES is to designate the IP address of AGN Networks SIP Proxy as a trusted host. As a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the designated IP address.[2]

If multiple SIP proxies are used, the IP address of each SIP proxy must be added as a trusted host.

To configure a trusted host:

- Telnet to the Avaya SES IP address (i.e., in this case 10.1.1.86) and log in using the administrative login and password.

- Enter the following trustedhost command at the Linux shell prompt:

  ```
  trustedhost –a 20.1.1.54 –n k2.devcon.com –c AGN_Procy
  ```

  The –a argument specifies the address to be trusted; –n specifies the SES host name; –c adds a comment.

- Use the following trustedhost command to verify the entry is correct:

  ```
  trustedhost –L
  ```

  Figure 31 illustrates the results of the trustedhost commands.[3]

- Complete the trusted host configuration by returning to the main Avaya SES Administration web page and again clicking on the **Update** link as shown in **Figure 30**.

  If the **Update** link is not visible, refresh the page by selecting **Top** from the left hand menu. Note this step is required even though the trusted host was configured via the Linux shell.

[2] Note, if the trusted host step is not done, authentication challenges to incoming SIP messages (such as INVITEs and BYEs) will be issued but not responded to. This may cause call setup to fail, active calls to be disconnected after timeout periods, and/or SIP protocol errors.

[3] For completeness, the –d argument allows the trust relationship to be deleted. For, example,

```
trustedhost –d 20.1.1.54 –n k2.devcon.com
```

removes the trust relationship added above.

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```
admin@k2> trustedhost -a 20.1.1.54 -n k2.devcon.com –c AGN_Proxy
20.1.1.54 is added to trusted host list.

admin@k2> trustedhost -L
Third party trusted hosts.
       Trusted Host       |       CCS Host Name       |         Comment
--------------------------+---------------------------+-------------------------
20.1.1.54                 | k2.devcon.com             |    AGN_Proxy
```

Figure 31: Configuring a Trusted Host

Important Note: After making any configuration changes on Avaya SIP Enablement Services, the user must click on the **Update** link in the Administration web interface for the changes to take effect.

4.2. Configuration for SIP Telephones

This section provides very basic instructions for completing the administration necessary to support the optional Avaya 46xx SIP telephones. Additional features such as the use of mnemonic addressing and instant messaging are also supported by Avaya SES but are beyond the scope of these Application Notes.

Step 1: Add a SIP User
Create the SIP user record as follows:

- In Avaya SES administration, expand the **Users** link in the left side blue navigation bar and click on the **Add** link.
- In the **Add User** screen, enter the extension of the SIP endpoint in the **Primary Handle** field.
- Enter a user password in the **Password** and **Confirm Password** fields. This password will be used when logging into the user's SIP telephone.
- In the **Host** field, select the Avaya SIP Enablement Services server hosting the domain (*devcon.com*) for this user. Enter the **First Name** and **Last Name** of the user.
- To associate a media server extension with this user, select the **Add Media Server Extension** checkbox. Calls from this user will always be routed through Avaya Communication Manager over the SIP trunk for origination services.
- Press the **Add** button. This will cause a confirmation screen to appear.
- Press **Continue** on the confirmation screen.

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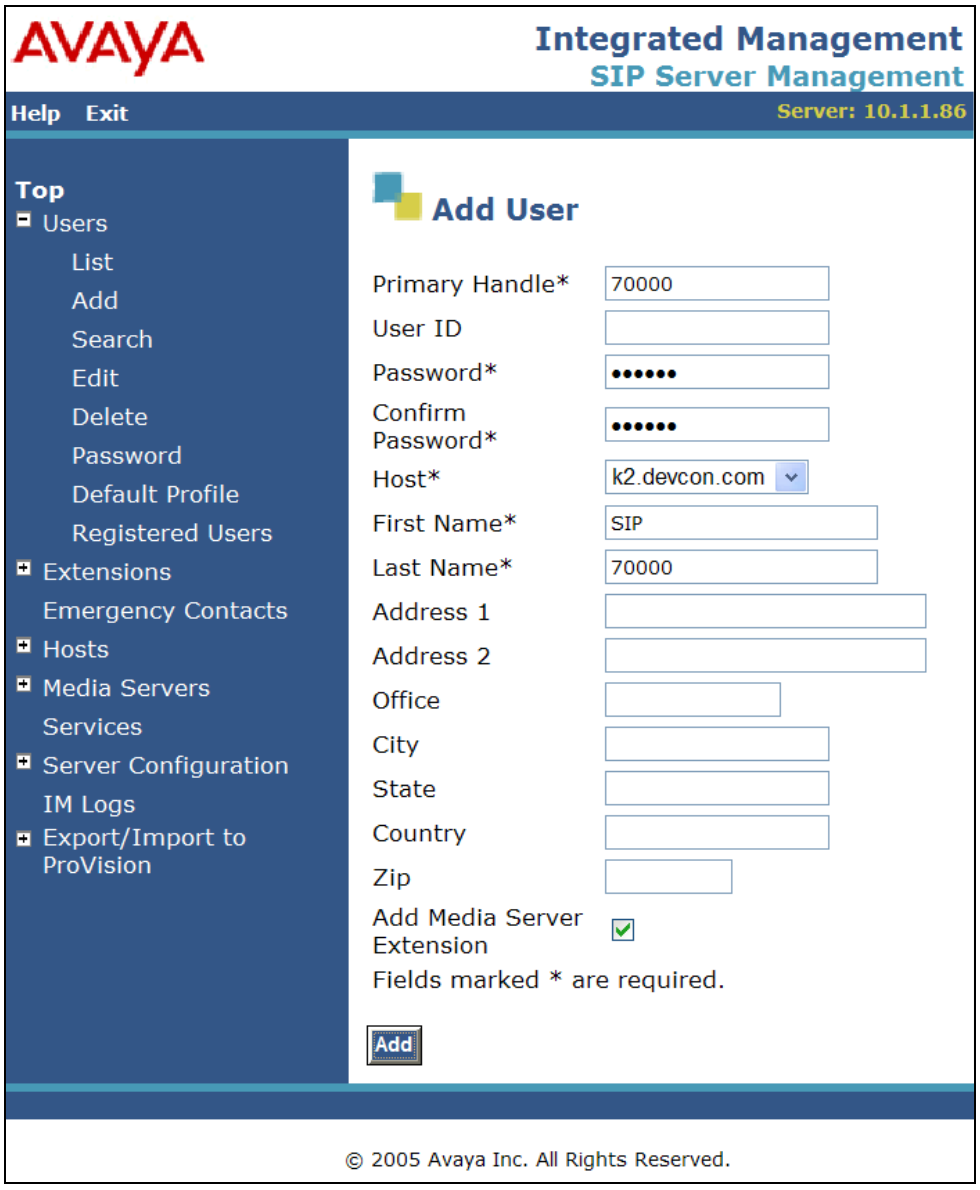


Figure 32: Add User

Step 2: Specify Corresponding Avaya Communication Manager Extension

The SIP phone handle must now be associated with the corresponding extension on Avaya Communication Manager.

- In the **Add Media Server Extension** screen, enter the **Extension** configured on the media server, shown in **Figure 15**, for the OPS extension on Avaya Communication Manager previously defined in Section 3.2. Usually, the media server extension and the user extension are the same (recommended) but it is not required to be.
- Select the **Media Server** assigned to this extension.
- Click on the **Add** button.
- To commit the configuration changes, click on the **Update** link in the left pane.

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Figure 33: Add Media Server Extension

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

5. AGN Networks On Demand SIP Service Configuration

In order to use the On Demand SIP service, a customer must request service using the AGN Networks sales process. The process can be started by contacting AGN Networks via the links found on their corporate web site at http://agnvoip.com and requesting information via the sales links or telephone numbers.

During the signup process, AGN Networks will require that the customer provide the public IP address used to reach the Avaya SIP Enablement Services server. (Note the address used within these Application Notes is 10.1.1.86; the actual IP address will be specific to the customer implementation).

Following signup, AGN Networks will provide the following:

- Username and Password to access the customer support web site.
- IP address of the AGN Networks SIP Proxy Server.

Once this information is available, the remaining configuration is performed using a web browser with Internet access to the AGN Networks web site.

Step 1: Access the AGN Networks Web Site
To begin, access the AGN Networks web site at http://www.agnvoip.com and click on the **USER LOGIN** link in the top navigation bar as shown in **Figure 34**. The my.agnnetworks.com login page will appear.

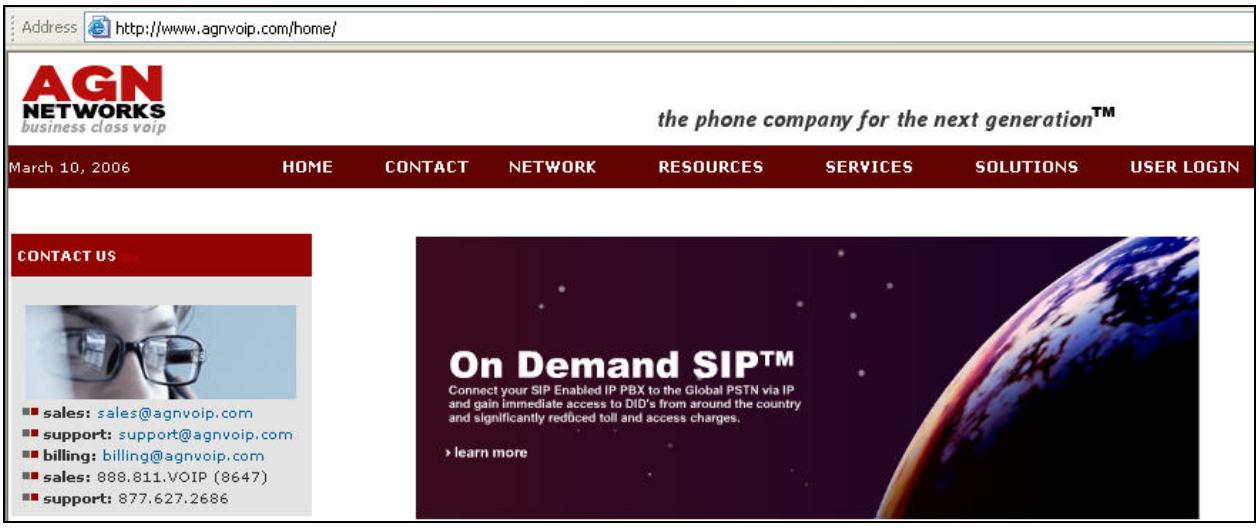


Figure 34 - AGN Networks Home Page

Step 2: Log In
Log in by entering the **Username** and **Password** provided by AGN Networks as shown in **Figure 35.**

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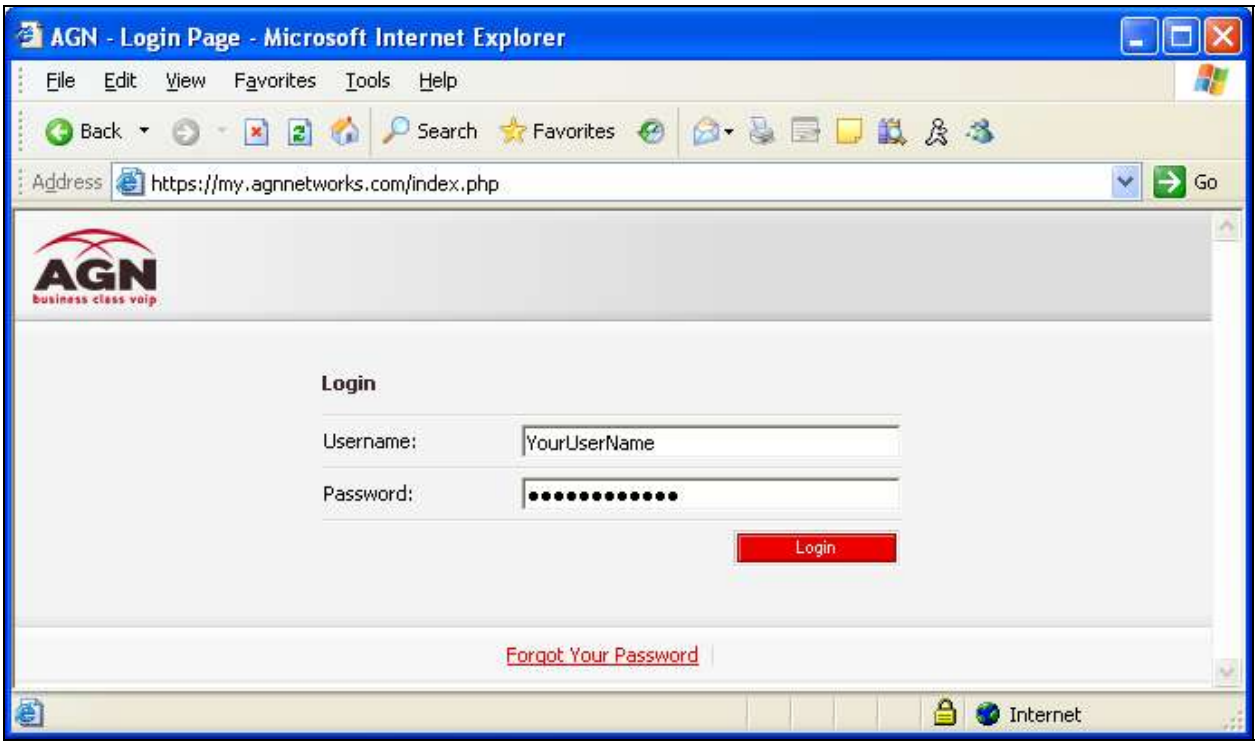


Figure 35 - AGN Networks Customer Login Screen

Step 3: Add a New Location

After login, select the **Locations** tab in the horizontal navigation bar to display the **Location List** page as shown in **Figure 36**. Click on the **Add Location** link found in the left hand menu.



Figure 36 - AGN Networks Locations Screen

Step 4: Specify the Location Information

On the Add Location page **(Figure 37)** enter the following:

- **Location Name** – Enter an alphabetic name describing the location that theAvaya SES edge server supports or where the server resides. In this example, "Headquarters" is used as the location name.

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(Note that there should only be one location entry (corresponding to the SES edge server) regardless of the number of customer sites served by individual Avaya Communication Managers.)

- **Extension** – Any extension number within the location. Typically this would correspond to the Listed Directory Number for the location. In this example, "7000" is entered.
- **IP Address** – The public IP address used to reach the Avaya SES server. In this example, 10.1.1.86 is used.
- **VoIP Protocol** – Choose SIP.
- **Location Address** – Enter the US postal address for the location.

Press the **Submit** button to save the information.

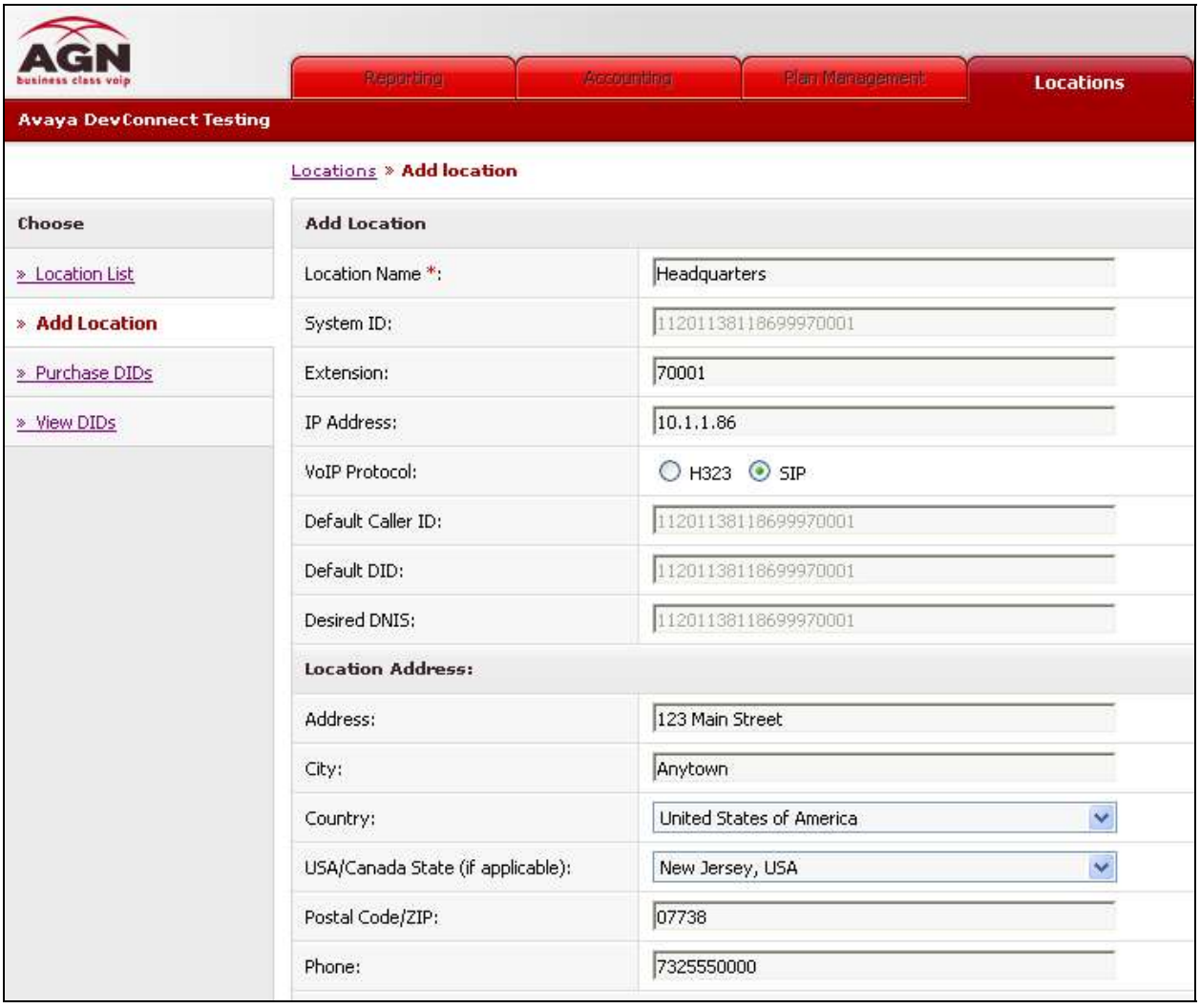


Figure 37 - AGN Networks Add Location Form

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Step 5: Obtain DID and/or TollFree Numbers
To receive incoming calls, either DID and/or TollFree numbers must be purchased.

- From the **Locations** tab, click the **Purchase DIDs** link found on the left hand menu. The form shown in **Figure 38** appears.
- For DID numbers, select the desired **State**, **Area Code** and **City** from the drop down lists as shown.
- Enter the number of DID numbers (for that City) desired into the **Desired Quantity** field.
- Press the **Submit** button.
- Click **Yes** to indicate acceptance of additional charges in the pop-up window that may appear.

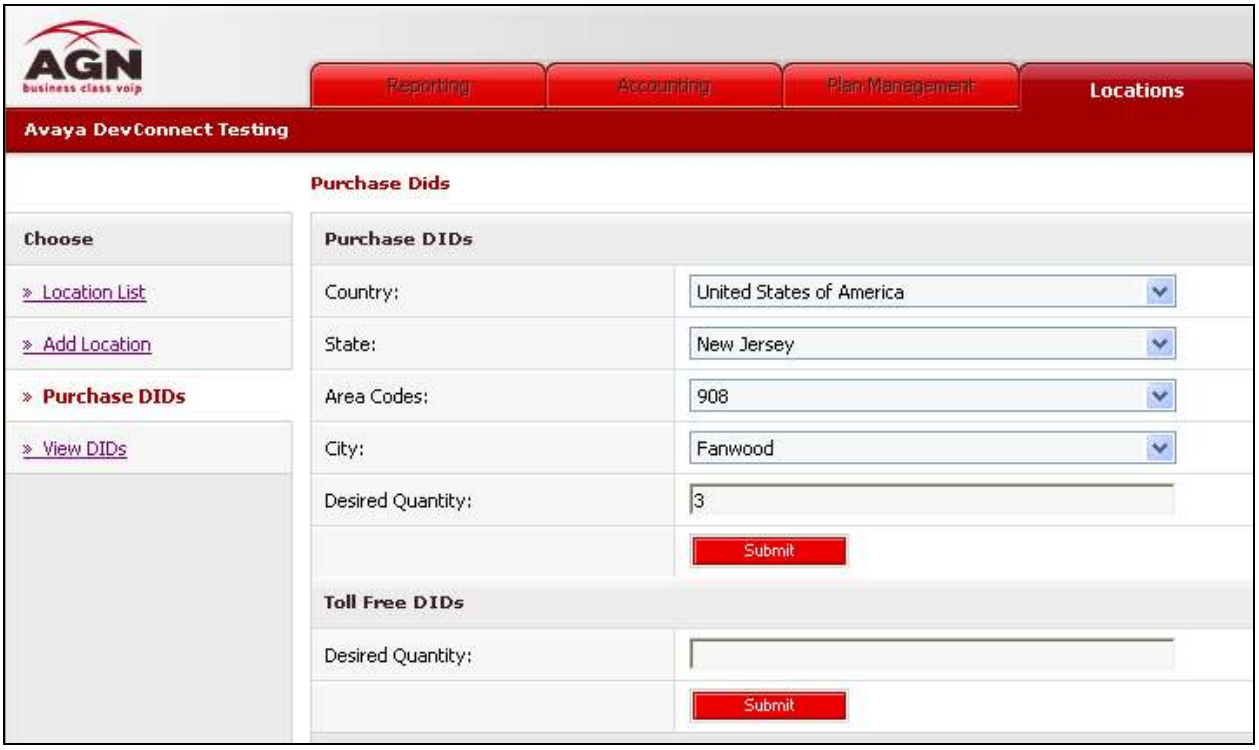


Figure 38 - AGN Networks Purchase DIDs Form

- Following the submission, use the **View DIDs** link to see the DID numbers reserved for your use as shown in **Figure 39**.

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Figure 39 – AGN Networks Reserved DID Numbers

Step 6: Assign DID Numbers

The reserved DID numbers must now be assigned to the Headquarters location previously defined.

- Choose **Location List** from the left menu.
- Click on the link for the name of the location (i.e., Headquarters). The **Update Location** form **(Figure 36)** previously completed will appear.



Figure 40 - AGN Networks Location Screen

- Scroll to the bottom on that form and select the **Manage DIDs** link located just above the **Submit** button **(Figure 41)**.



Figure 41 - AGN Networks Location Form Manage DIDs Link

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The **Managed DIDs** link will open a page as shown in **Figure 42**.

- Select one or more DID number(s) from the **Reserved DIDs** list and use the >> button to move the numbers to the **Assigned DIDs** list.
- Repeat as necessary for other DID numbers.
- Press the **Submit** button and confirm that the DIDs were successfully assigned to that location **(Figure 43)**.

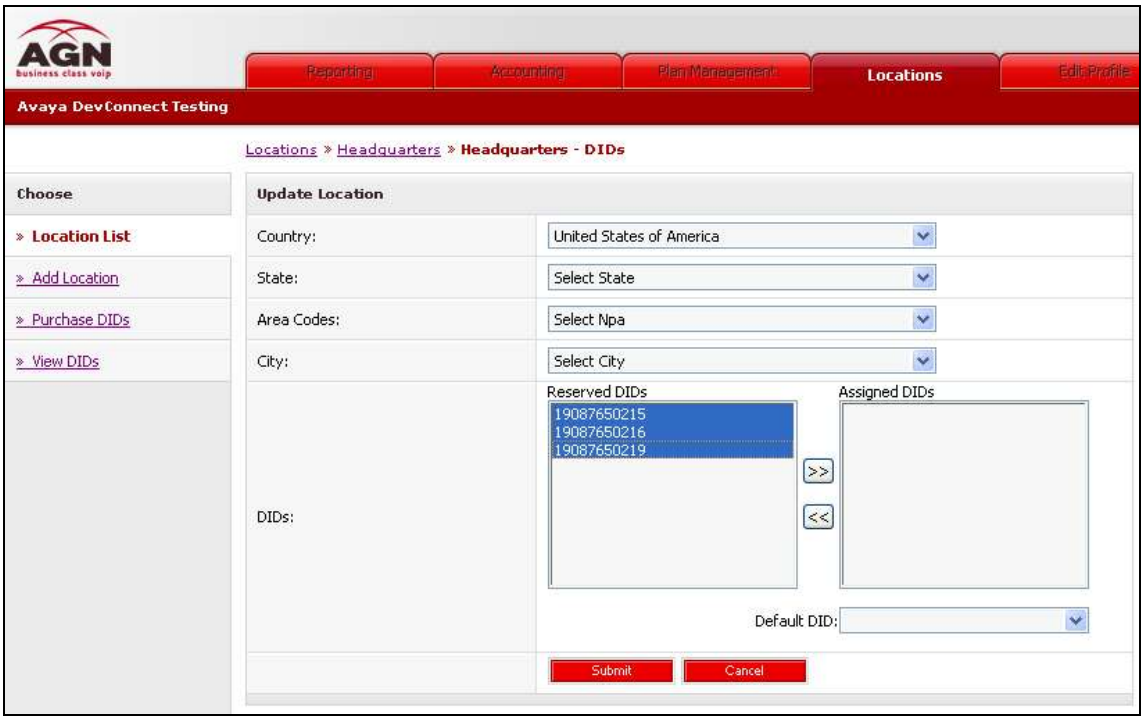


Figure 42 - Assign DIDs Form

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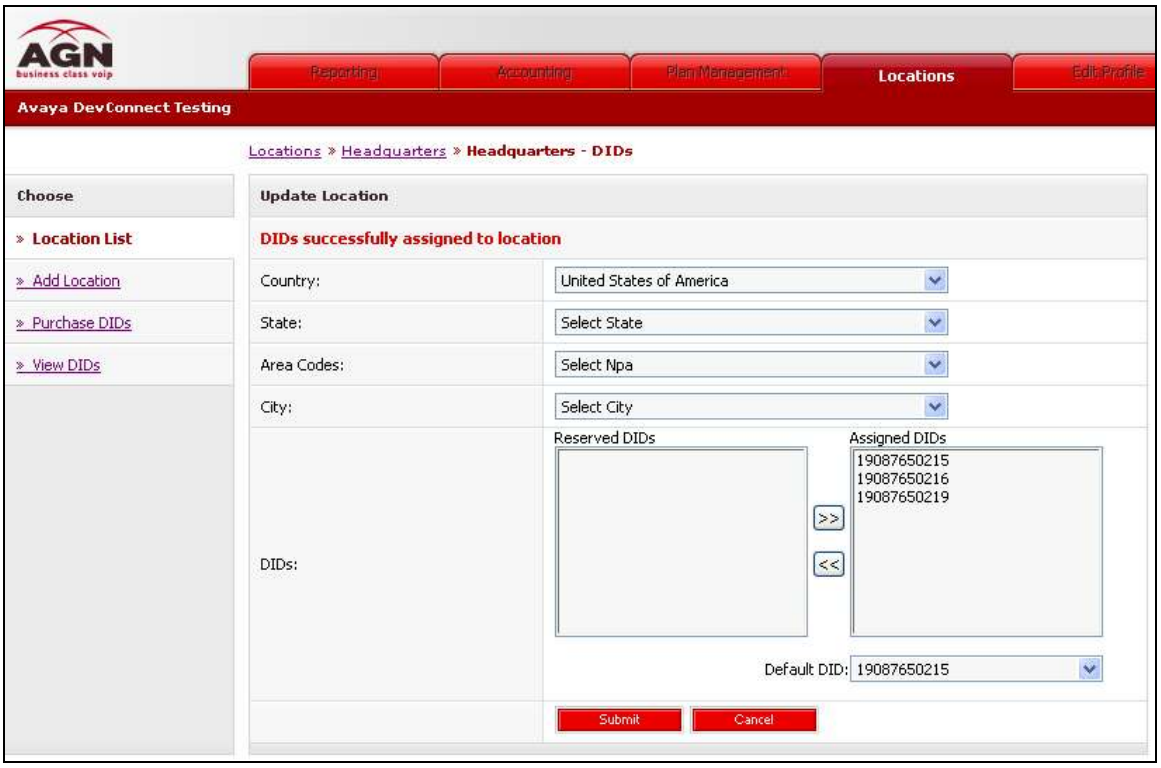


Figure 43 - Successfully Assigned DIDs

Step 7: Specify the Desired DNIS Values

Finally, update the Desired DNIS values to match the routing strategy defined in Section 4 during the configuration of Avaya SES. Recall that the numbering strategy was to define the DNIS value to be a 3-digit routing number followed by the full 11 digit North American telephone number.

- Begin by choosing the **View DIDs** link found on the left hand menu of the **Locations** tab. A screen similar to **Figure 44** below will be seen. Note that the current **Desired DNIS** values do not currently match the desired numbering strategy.

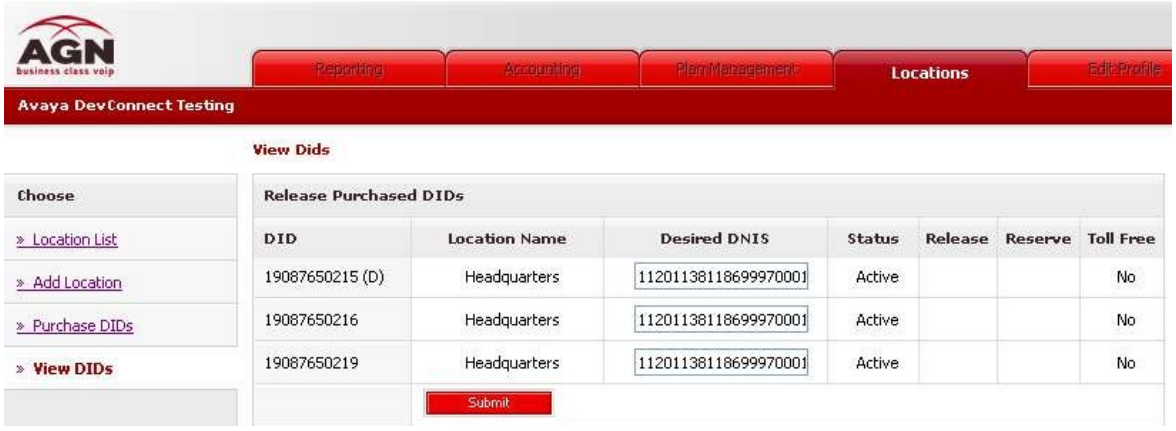


Figure 44 - View DIDs Form showing Desired DNIS Values

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- Update each **Desired DNIS** field accordingly and press the **Submit** button. In this example, the Desired DNIS values should be "901" followed by the DID number shown in the left column.
- Verify that the correct Desired DNIS values are now recorded and the DID status is Active as shown in **Figure 45**.

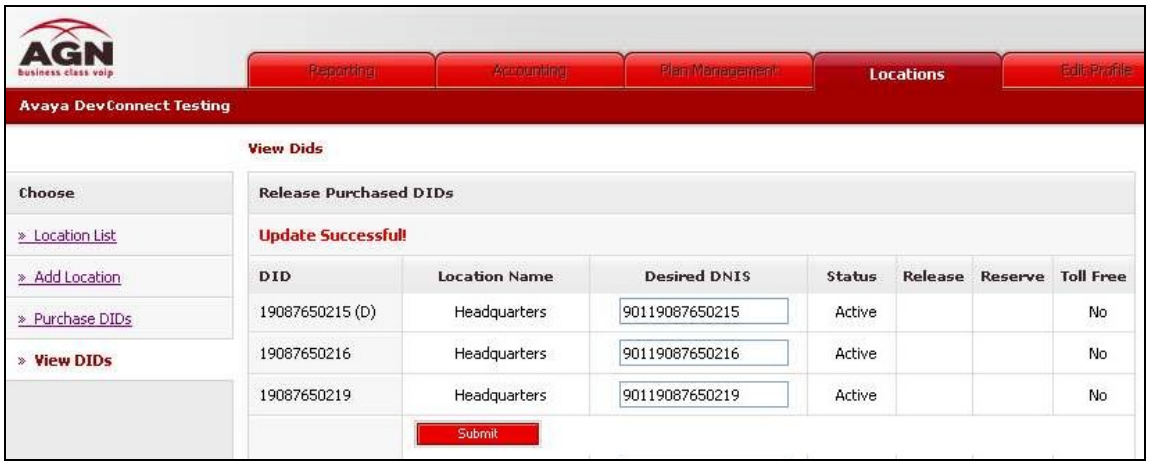


Figure 45 – Successful Update of Desired DNIS fields.

This completes the configuration within AGN Networks for inbound and outbound calling.

6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the AGN Networks On Demand SIP services and an Avaya SIP-based configuration. This section covers the general test approach and the test results.

6.1. General Test Approach

A simulated enterprise site consisting of an Avaya IP telephony 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 On Demand SIP services provided by AGN Networks. This allowed the enterprise site to use SIP trunking for PTSN calling.

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 AGN Networks.
- Outgoing calls from the enterprise site to the PSTN.
- Calls using SIP, H.323, digital and analog endpoints supported by the Avaya IP telephony solution.
- Various call types including: local, long distance, international, toll free, operator and directory assistance calls.

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- Calls using G.711 and G.729A codecs.
- DTMF transmission using RFC 2833.
- Voicemail coverage and retrieval for endpoints at the enterprise site.
- Direct IP-to-IP media (also known as "Shuffling") which allows SIP endpoints to send audio (RTP) packets directly to each other without using media resources on the Avaya Media Gateway.

6.2. Test Results

All test cases passed.

7. Verification Steps

This section provides verification steps that may be performed in the field to verify that the SIP, H.323, digital and analog endpoints can place outbound and receive inbound PSTN through the AGN Networks services.

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
2. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.
3. Verify that the user on the PSTN can terminate an active call by hanging up.
4. Verify that an endpoint at the enterprise site can terminate an active call by hanging up.
5. If the Direct-IP feature (a.k.a. Shuffling) is enabled, verify that a call originated or terminated on an Avaya 4600 Series SIP Telephone has the RTP path directly between the SIP phone and AGN Networks gateway. To determine if the call is shuffled, identify the trunk member active on the call by running the **status trunk <*group*>** command using the SAT of Avaya Communication Manager. Next, run the **status trunk group/member** command and check the **Audio Connection** field. If the call is shuffled, the field should be set to *ip-direct*; otherwise, the field would be set to *ip-tdm*.

8. Support

For technical support on AGN Networks On Demand SIP Services, contact support at 1-888-811-8647 or support@agnvoip.com.

9. Conclusion

These Application Notes describe the configuration steps required to connect an enterprise site consisting of an Avaya SIP-based telephony solution to the AGN Networks On Demand SIP Services. SIP trunking permits enterprise customers to reduce long distance and interconnection costs by using broadband Internet access to support PSTN telephony. In addition, customers can benefit from additional features such as web-based provisioning, virtual DID numbers and online reports available as part of the On Demand SIP service.

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10. References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at http://support.avaya.com.

[1] *Administrator Guide for Avaya Communication Manager*, June 2005, Issue 1, Document Number 03-300509.

[2] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, June 2005, Issue 9, Document Number 210-100-500.

[3] *Converged Communications Server R3.0 Installation and Administration Guide (SIP Enablement Services R3.0)*, June 2005, Issue 5, Document Number 555-245-705.

[4] *IA 770 INTUITY AUDIX Release 3.0 Installation, Upgrades, and Troubleshooting*, April 2005, Issue 1, Document Number 11-300532.

[5] *SIP Support in Release 3.0 of Avaya Communication Manager Running on the Avaya S8300, S8500, S8500B, S8700, and S8710 Media Server*, June 2005, Issue 5, Document Number 555-245-206.

[6] *4600 Series IP Telephone R2.3 LAN Adminstrator Guide*, November 2005, Document Number 555-233-507

Additional information about the AGN Network Services is available at http://www.agnvoip.com.

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APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by the AGN Networks 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 AGN Networks to Avaya SIP Enablement Services:

```
INVITE sip:90112014710200;npdi=yes@10.1.1.86 SIP/2.0
Max-Forwards: 68
Session-Expires: 3600;Refresher=uac
Supported: timer
To: <sip:12014710200@20.1.1.54:5060>
From: <sip:7328521638@20.1.1.54:5060;GX=GXAVATELDIDGOLD-
5ed61h7hbtmva>;tag=SD40ivd01-gK0a27a357
Call-ID: 147-3340972610-679906@agnnt01.avatel.com
CSeq: 1 INVITE
Via: SIP/2.0/UDP
20.1.1.54:5060;branch=z9hG4bK20eddeef31453f7486e6a1c7392c1c6c
Contact: sip:7328521638@20.1.1.54:5060
Content-Type: application/sdp
Content-Length: 299

v=0
o=NexTone-MSW 22039 12517 IN IP4 20.1.1.54
s=sip call
c=IN IP4 20.1.1.54
t=0 0
m=audio 30570 RTP/AVP 18 0 8 100
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:100 telephone-event/8000
a=fmtp:100 0-15
a=sendrecv
a=maxptime:20
```


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Sample SIP INVITE Message from Avaya SIP Enablement Services to AGN Networks:

```
INVITE sip:17328521638@20.1.1.54:5060;transport=udp SIP/2.0
Call-ID: 0405c5fd660da12470439197cf00
CSeq: 2 INVITE
From: "70000" <sip:70000@10.1.1.86>;tag=0405c5fd660da12370439197cf00
Record-Route:
<sip:10.1.1.86:5060;transport=udp;lr>,<sip:10.1.1.84:5061;lr;transport=tls>
To: sip:17328521638@devcon.com
Via: SIP/2.0/UDP
10.1.1.86:5060;branch=z9hG4bK030303434343030303035b58.0,SIP/2.0/TLS
10.1.1.84;psrrposn=2;branch=z9hG4bK0405c5fd660da12670439197cf00,SIP/2.0/UDP
10.1.1.72:5060;psrrposn=1;branch=z9hG4bK8acf119c3
Content-Length: 160
Content-Type: application/sdp
Contact: "70000" <sip:7327770000@10.1.1.84:5061;transport=tls>
Max-Forwards: 67
User-Agent: Avaya CM/R013x.00.0.340.3
Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
Session-Expires: 600;Refresher=uac
Min-SE: 600
Supported: 100rel,timer,replaces,join
P-Asserted-Identity:"SIP70000"<sip:7327770000@devcon.com:5061>

v=0
o=- 1 1 IN IP4 10.1.1.84
s=-
c=IN IP4 10.1.1.85
t=0 0
m=audio 2442 RTP/AVP 0 127
a=rtpmap:0 PCMU/8000
a=rtpmap:127 telephone-event/8000
```


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APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within the Avaya SES is a Linux regular expression used to match against the URI string found in the SIP INVITE message.

Regular expressions are a way to describe text through pattern matching. The regular expression is a string containing a combination of normal text characters, which match themselves, and special *metacharacters*, which may represent items like quantity, location or types of character(s).

In the pattern matching string used in the Avaya SES:

- Normal text characters and numbers match themselves.
- Common metacharacters used are:
  - A period **.** matches any character once (and only once).
  - A asterisk ***** matches zero or more of the preceding characters.
  - Square brackets enclose a list of any character to the 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.
  - Curley brackets containing an integer 'n' indicate that the preceding character must be matched exactly 'n' time. 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:17328521638@20.1.1.54:5060;transport=udp SIP/2.0
```


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