Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the PAETEC Communications IPATH Service and an Avaya SIP Telephony Solution – Issue 1.0

Abstract

These Application Notes describe the steps to configure SIP trunking between the PAETEC Communications IPATH Service and an Avaya SIP Telephony solution consisting of Avaya Communication Manager, Avaya SIP Enablement Services, and various Avaya endpoints.

PAETEC Communications, Inc., is a supplier of communication solutions to medium and large businesses and institutions. They offer personalized solutions that include a comprehensive suite of Voice over Internet Protocol (VoIP) services delivered over their Private-IP Multi-Protocol Label Switching (MPLS) network. SIP trunking to the PAETEC Communications IPATH Service allows customer locations to be connected to the public telephone network via converged IP network access serving both voice and data needs. This provides a flexible, cost-saving alternative to traditional hardwired telephone trunk lines.

PAETEC Communications 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.

1. Introduction

These Application Notes describe the steps for configuring SIP trunking between the PAETEC Communications IPATH Service and an Avaya SIP 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 one-X Desktop Edition running on a Microsoft Windows PC.

PAETEC Communications currently serves customers in 29 markets. An illustration of PAETEC Communications network coverage is provided in **Figure 1.** An updated interactive version of this map may be found online at http://www.paetec.com/2_1/2_1_5__2.html.

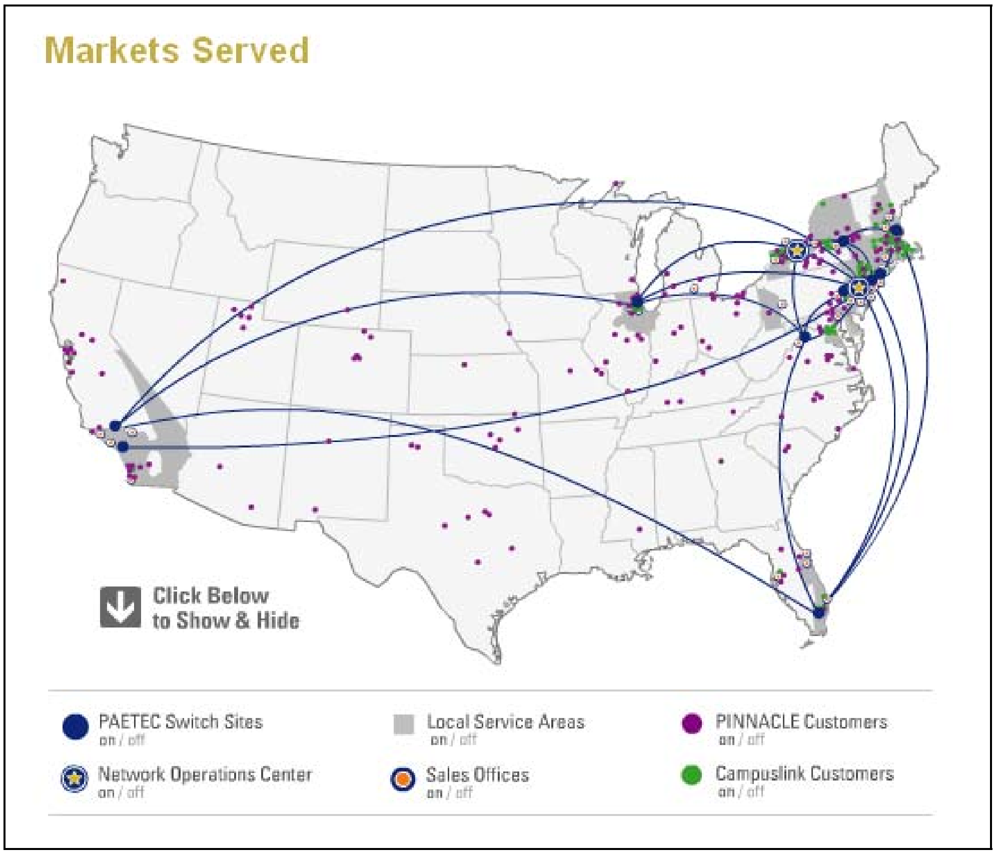

Figure 1 – PAETEC Communications Market Coverage

These Application Notes describe the configuration of SIP trunking to access the PAETEC Communications IPATH Service. Access to the PAETEC Communications service supports the following capabilities:

- Outbound PSTN calling to local, long distance and international services
- Incoming Direct Inward Dial (DID) service
- Incoming Toll-free service
- Operator, Directory Assistance and Calling Card Service
- Converged IP access via a private IP MPLS Network.

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

Figure 2 illustrates an example Avaya SIP Telephony solution connected to PAETEC Communications IPATH Service using SIP trunking. This is the configuration for the compliance testing.

The Avaya SIP Telephony solution used to create a simulated customer site contained:

- An Avaya S8710 Media Server with an Avaya G650 Media Gateway. The S8710 served as the host processor for Avaya Communication Manager.
- 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.

Although not shown in **Figure 2**, the enterprise site may also have alternate routes to the PSTN using traditional trunks.

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

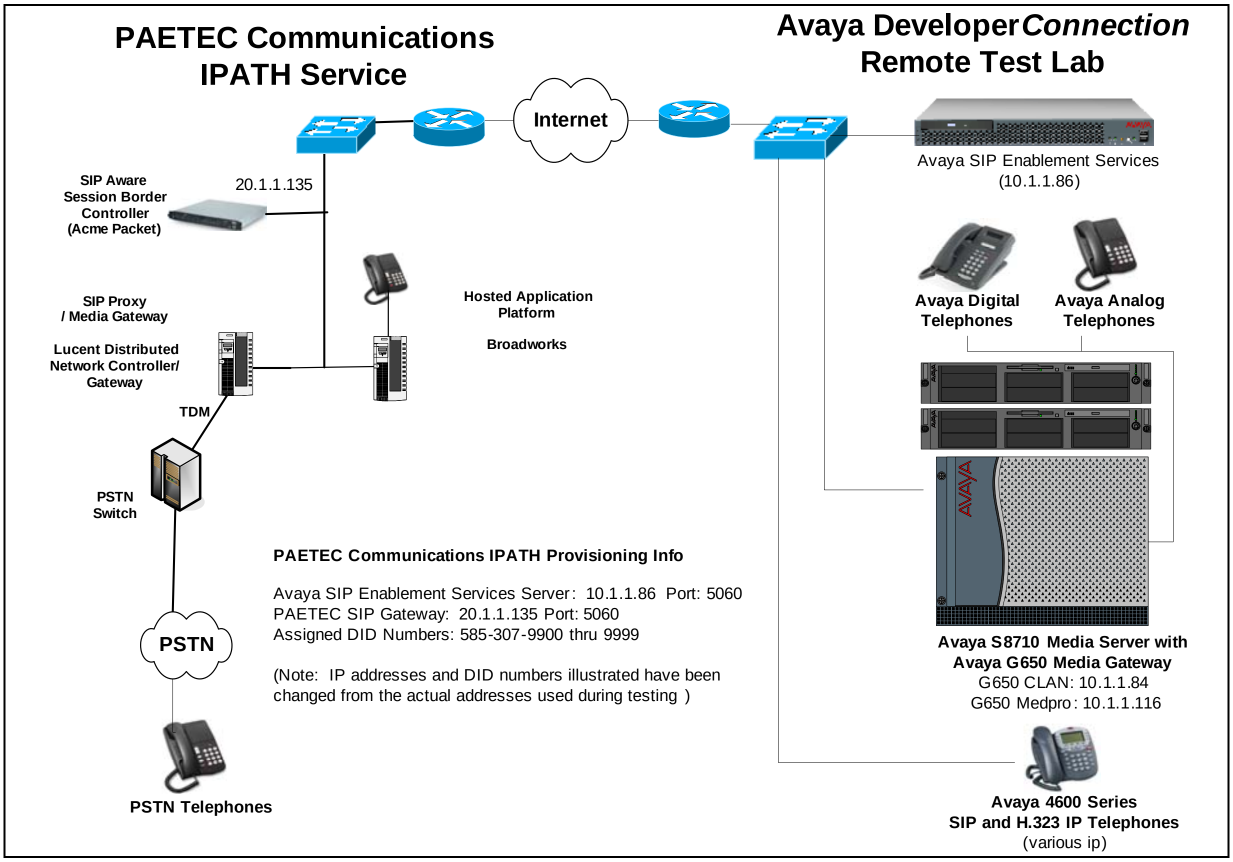

Figure 2: Avaya SIP Telephony Network using PAETEC Communications IPATH Service

1.1. Call Flows

To better understand how calls are routed between the PSTN and the enterprise site shown in **Figure 2** using SIP trunks, two call flows are described in this section.

The first call scenario illustrated in **Figure 3** 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 PAETEC Communications provided DID number assigned to an Avaya Communication Manager telephone at the enterprise site. The PSTN routes the call to the PAETEC Communications network (as the local service provider) who routes the DID number to the assigned customer.

2. Based on the DID number, PAETEC Communications offers the call to Avaya SES using SIP signaling messages sent over the converged access facility. Note that the assignment of the DID number and the address of the Avaya SES server was previously established during the ordering and provisioning of the service.

3. Avaya SES routes the call to the Avaya S8710 Media Server running Avaya Communication Manager over a SIP trunk.

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 telephones.

   \- or –

4a. Inbound calls destined for a SIP extension at the enterprise are routed to Avaya Communication Manager. The Avaya Communication Manager (acting as a SIP back-to-back user agent) then transmits the appropriate SIP signaling via Avaya SES to the SIP telephone (as shown by the 4a arrow.)

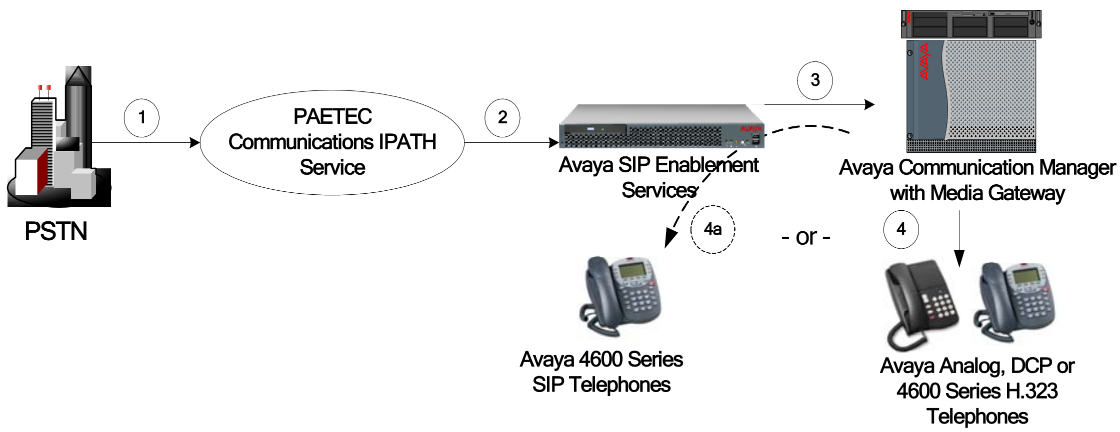

Figure 3: Incoming PSTN Calls to Avaya Communication Manager

Appendix A illustrates an example of a SIP INVITE message sent by PAETEC Communications for an incoming DID call.

The second call scenario illustrated in **Figure 4** is an outgoing call from an Avaya telephone at the enterprise site to the PSTN via the SIP trunk to PAETEC Communications.

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 PAETEC Communications.

4. PAETEC Communications completes the call to the PSTN.

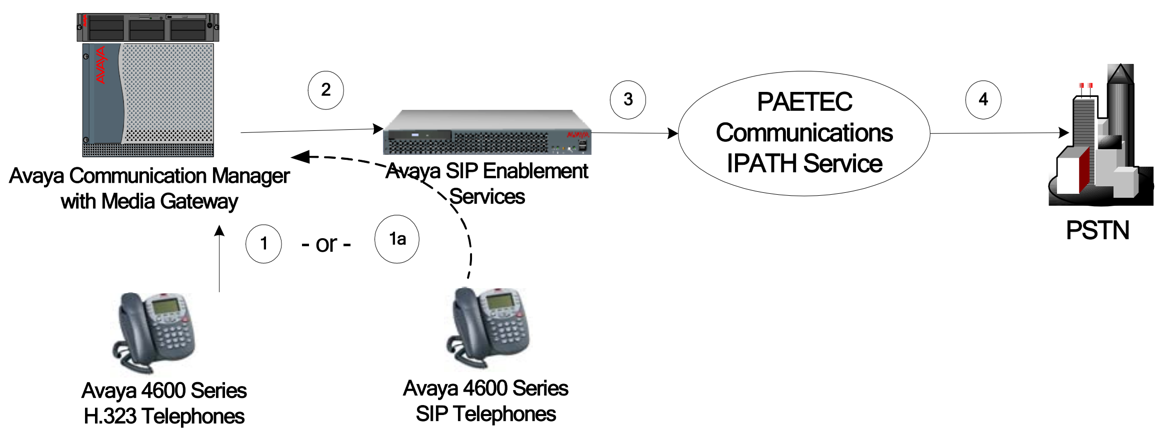

Figure 4: Outgoing Calls from Avaya Communication Manager to the PSTN

2. Equipment and Software Validated

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

Avaya SIP Telephony Solution Components	
Hardware Component	**Version**
Avaya S8710 Media Server with an Avaya G650 Media Gateway	Communication Manager 3.1 (R013x.01.0.628.6-11410)
Avaya SIP Enablement Services on S8500B Media Server	SES-3.0.0.0—031.0
Avaya 4620SWSeries SIP Telephones	Release 2.2.2
Avaya one-X Desktop Edition	Release 2.1 Build 44
Avaya 4620SW Series H.323 IP Telephones	Release 2.3
Avaya 6416 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a

PAETEC Communications IPATH Service Components	
Hardware Component	**Version**
Lucent Distributed Network Controller/Gateway	5.1.0.3 SP 16
Acme Packet Net-Net Session Border Controller	2.1.0 P41
Broadworks Hosted Platform	12.0

The specific configuration above was used for the PAETEC Communications compatibility testing. Note that this solution will be compatible with all other Avaya Media Server and Media Gateway platforms running similar versions of Avaya Communication Manager and Avaya SIP Enablement Services.

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 server. This trunk will carry the SIP signaling sent to the PAETEC Communications IPATH Service.

This SIP trunk also provides the trunking for SIP endpoint devices such as Avaya 4600 SIP telephones and Avaya one-X Desktop Edition using Avaya Communication Manager in the recommended Off-PBX Station (OPS) configuration. Avaya SIP telephones are configured as OPS stations on Avaya Communication Manager. OPS SIP 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.

Note the use of SIP endpoints is optional. The steps discussed in Sections 3.2 and 4.2 describing SIP endpoints administration may be omitted if SIP endpoints are not used.

In the Avaya SIP architecture, the Avaya SES acts as a SIP proxy through which all incoming and outgoing SIP messages flow to PAETEC Communications. There is no direct SIP signaling path between PAETEC Communications and Avaya Communication Manager or Avaya SIP endpoints.

For incoming calls, the Avaya SES uses media server routing maps to direct the incoming SIP messages to the appropriate Avaya Communication Manager. Once the message arrives at the Avaya Communication Manager, further incoming call treatment such as incoming digit translations, class of service restrictions, etc. may be performed.

All outgoing calls to the PSTN are processed within Avaya Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Avaya Communication Manager selects a SIP trunk, the SIP signaling is routed to the Avaya SES. Within the Avaya SES, host address maps direct the outbound SIP messages to the PAETEC Communications SIP gateway.

The dial plan for the configuration described in these Application Notes consists 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 routes all calls using Automatic Route Selection (ARS), except for intra-switch calls. The configuration of ARS is beyond the scope of these Application Notes and the reader should refer to [1] and [2] for additional information.

The Avaya Communication Manager configuration was performed using the System Access Terminal (SAT). The general installation of the S8710 media server, G650 Media Gateway and circuit packs such as the CLAN is presumed to have been previously completed and is not discussed 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 5**. 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 assigned. Maximum Off-PBX Telephones indicates the maximum number available in the system and is controlled by the license file. The USED column indicates the number of telephones licenses currently assigned. The difference between the two represents the additional number of SIP endpoints that can be added.

```
display system-parameters customer-options                     Page   1 of  10
                                OPTIONAL FEATURES

     G3 Version: V13
       Location: 1                            RFA System ID (SID): 1
       Platform: 8                            RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 44000 86
                                      Maximum Stations: 36000 36
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 0     0
                    Maximum Off-PBX Telephones -   OPS: 100   17
                    Maximum Off-PBX Telephones - SCCAN: 0     0

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

Figure 5: 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 the PAETEC Communications network, SIP endpoints and any other SIP trunks used. Note that calls from non-SIP endpoints to PAETEC Communications will use one SIP trunk for the duration of the call. Each SIP OPS telephone on a call with PAETEC Communications will require two SIP trunks for the duration of the call.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 0     0
           Maximum Concurrently Registered IP Stations: 100   2
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 100   50

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 1     1
                    Maximum G250/G350/G700 VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 2     1
          Maximum TN2602 Boards with 320 VoIP Channels: 2     0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

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

Figure 6: 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 “CLAN” and “10.1.1.84” are the name and IP address assigned to the TN799DP CLAN card. The CLAN entry was previously created during the installation of the system. Note, in smaller media gateways such as an Avaya G350, the S8300 processor address (procr) is used as the SIP signaling interface instead of the CLAN interface.

```
change node-names ip                                                Page   1 of   1
                                  IP NODE NAMES
    Name              IP Address           Name              IP Address
CLAN                10 .1  .1  .84                            .   .   .
default             0  .0  .0  .0                              .   .   .
ipsi                10 .1  .1  .109                            .   .   .
medpro-hw11         10 .1  .1  .116                            .   .   .
procr                  .   .   .                               .   .   .
SES                 10 .1  .1  .86                             .   .   .
val1-tn2501ap       10 .1  .1  .122                            .   .   .
                       .   .   .                               .   .   .
                       .   .   .                               .   .   .
( 11 of 11   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

Figure 7: IP Nodes Names Form

Step 3: Define IP Network Region

The **IP Network Region** form specifies the parameters used by the SIP trunk group serving the Avaya SES proxy (used to reach PAETEC Communications and any optional SIP endpoints). Note that these parameters also apply to any other elements (such as H.323 phones, media processor cards, etc.) also assigned to this region. In the **IP Network Region** form:

- 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 such as the TN2302AP IP Media Processor (MedPro) card.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this configuration, this codec set will apply to calls with PAETEC Communications 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 G650 Media Gateway, CLAN and MedPro cards. 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.

Note also that the **IP Network Region** form is used to set the packet parameters that provides priority treatment for signaling and audio packets over other data traffic on PAETEC Communications access facilities. These parameters may need to be aligned with the specific values provided by PAETEC 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 8: IP Network Region Form

Step 4: Define IP Codecs

Open the **IP Codec Set** form using the **Codec Set** value specified in the **IP Network Region** form (**Figure 8**) and enter the audio codec type to be used for calls routed over the SIP trunk. Typical settings of the **IP Codec Set** form are shown in **Figure 9.** 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 PAETEC Communications, only G.711MU, G.711A and/or G.729B can be included in this list.

```
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.729B           n          2        20
 2: G.711MU          n          2        20
 3:
 4:
```

Figure 9: IP Codec Set Form

Step 5: Configure the Signaling Group
Configure the **Signaling Group** form shown in **Figure 10** as follows:

- Set the **Group Type** field to *sip*.
- The **Transport Method** field will default to *tls* (Transport Layer Security). TLS is the only link protocol that is supported for SIP trunking with Avaya SIP Enablement Services.
- Specify the Avaya CLAN card (node name "CLAN") and the Avaya SIP Enablement Services Server (node name "SES") as the two ends of the signaling group in the **Near-end Node Name** and the **Far-end Node Name** fields, respectively. These field values are taken from the **IP Node Names** form shown in **Figure 7**. For smaller media server platforms, the near (local) end of the SIP signaling group may be the S8300 media server processor (procr) rather than the CLAN.
- 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 8**) into the **Far-end Network Region** field.
  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 remain 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
                                SIGNALING GROUP

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


   Near-end Node Name: CLAN                  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 10: Signaling Group Form

Step 6: Configure the Trunk Group
Configure the **Trunk Group** form as shown in **Figure 11** 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 10**.
- 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 PAETEC Communications. Calls involving a SIP endpoint and PAETEC Communications will use two trunk members for the duration of the call.

```
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 11: 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 12: 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 7xxxx should send the calling party number 585-307-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 13 shows the use of the "change public-unknown numbering" command to implement this rule.

```
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         58530          10
```

Figure 13: 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 sent in the INVITE message from PAETEC Communications are manipulated to route calls to the proper extension on Avaya Communication Manager. Note that this step must be consistent with the DID numbers and routing strategy defined in Sections 4.1 and 5.

In the examples used in these Application Notes, the incoming DID numbers provided by PAETEC Communications 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 14**:

- Open the **Incoming Call Handling Treatment** form for the SIP trunk group.
- For each extension assigned a DID number from PAETEC Communications, enter **10** into the **Called Len** and **Del** fields, and the entire **10 digit DID number** 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             10  5853079960        10   70000
 tie             10  5853079961        10   71001
 tie             10  5853079962        10   71002
```

Figure 14: Incoming Call Handling Treatment – Full Extension Mapping

If the customer's extension numbering plan aligns with the DID numbers (i.e., the final DID digits match the extension), it is not necessary to define an entry for each DID number. Assuming a PBX dial plan that used the 5 digit extensions 71000 thru 71999 and assuming PAETEC Communications provided DID numbers of 585-307-1000 thru 1999, the incoming number translation would be done similar to **Figure 15**. Note that the Called Number entry in this case represents the common matching portion applicable to all incoming numbers. Thus 5853071 matches all numbers in the assigned DID block from PAETEC Communications.

```
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              10    5853071         5
```

Figure 15: Incoming Call Handling Treatment – Simple Extension Mapping

Step 9: Save Avaya Communication Manager Changes
Enter the "save translation" command 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. SIP telephones are optional and not required to use the PAETEC Communications IPATH Service.

Step 1: **Assign a Station**
The first step in adding an off-PBX station for Avaya SIP telephones registered with Avaya SIP Enablement Services is to assign a station as shown in **Figure 16**.

Using the "add station" command from the SAT:

- Leave the station **Type** at the default "6408D+" value. (Note this is the Avaya recommended best practice that will prevent an alarm warning that occurs 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.

```
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 16: 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.

```
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 17: Station Administration – Page 2

On Page 3 of the **Station** form, configure at least 3 call appearances for the SIP telephone as shown in **Figure 18**.

```
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 18: Station Administration – 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) are described in Reference [6].

Step 2: **Configure Off-PBX Telephone 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 route the call to the SIP telephone.

On the **Off-PBX-Telephone Station-Mapping** form shown in **Figure 19**:

- 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 11**.
- Set the **Configuration Set** value. In these Application Notes, Configuration Set 1 uses the default values of the Configuration Set form.

```
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 19: 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 20: Stations with Off-PBX Telephone Integration – Page 2

Step 3: Repeat for each SIP Telephone
Repeat Steps 1 and 2 for each SIP telephone to be added.

Step 4: Save Avaya Communication Manager Changes
Enter the "save translation" command to make the changes permanent.

4. Configure Avaya SIP Enablement Services

This section covers the administration of Avaya SIP Enablement Services. 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. During the software installation, the install 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 [4].

This section is divided into two parts: Section 4.1 provides the steps necessary to configure SIP trunking to PAETEC Communications IPATH Service. Section 4.2 provides the steps necessary to complete the administration for optional SIP endpoints.

4.1. SIP Trunking to PAETEC Communications

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 server.

Log in with the appropriate credentials and then select the **Launch Administration Web Interface** link from the main screen as shown in **Figure 21**.

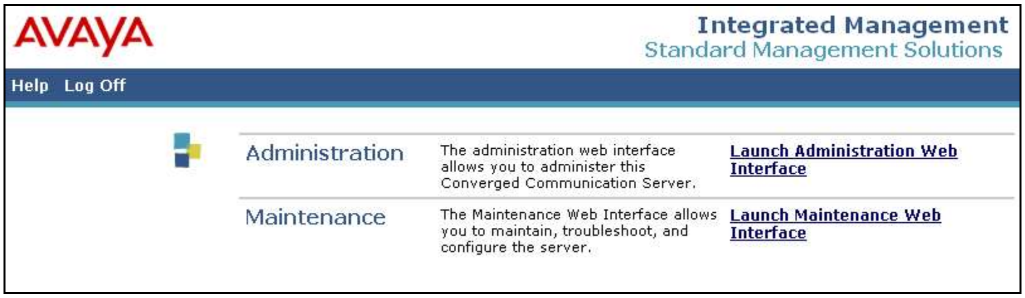

Figure 21 - Avaya SES Main Screen

The SES administration home screen shown in **Figure 22** should be displayed.

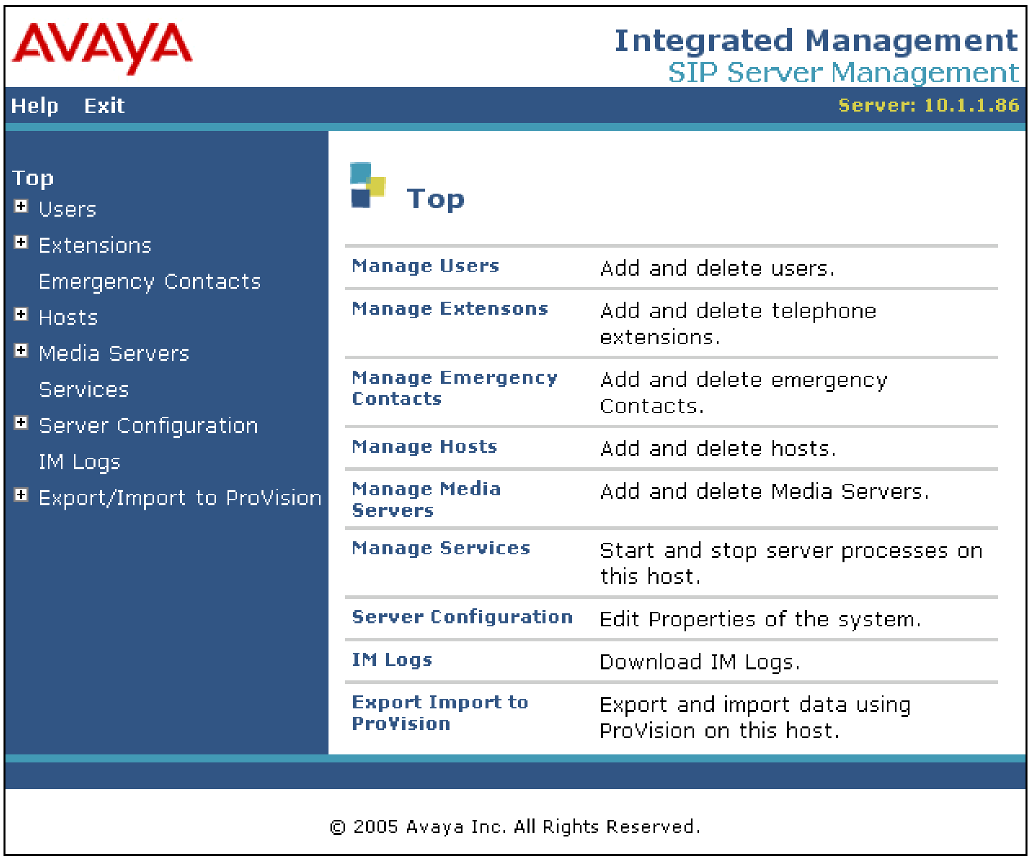

Figure 22: Avaya SES Administration Home Page

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 SES version and the network properties entered via the install script during the installation process.

In the **Edit System Properties** screen,

- Enter the **SIP Domain** name assigned to Avaya SIP Enablement Services.
- Enter a value into the **License Host** field. This value 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.

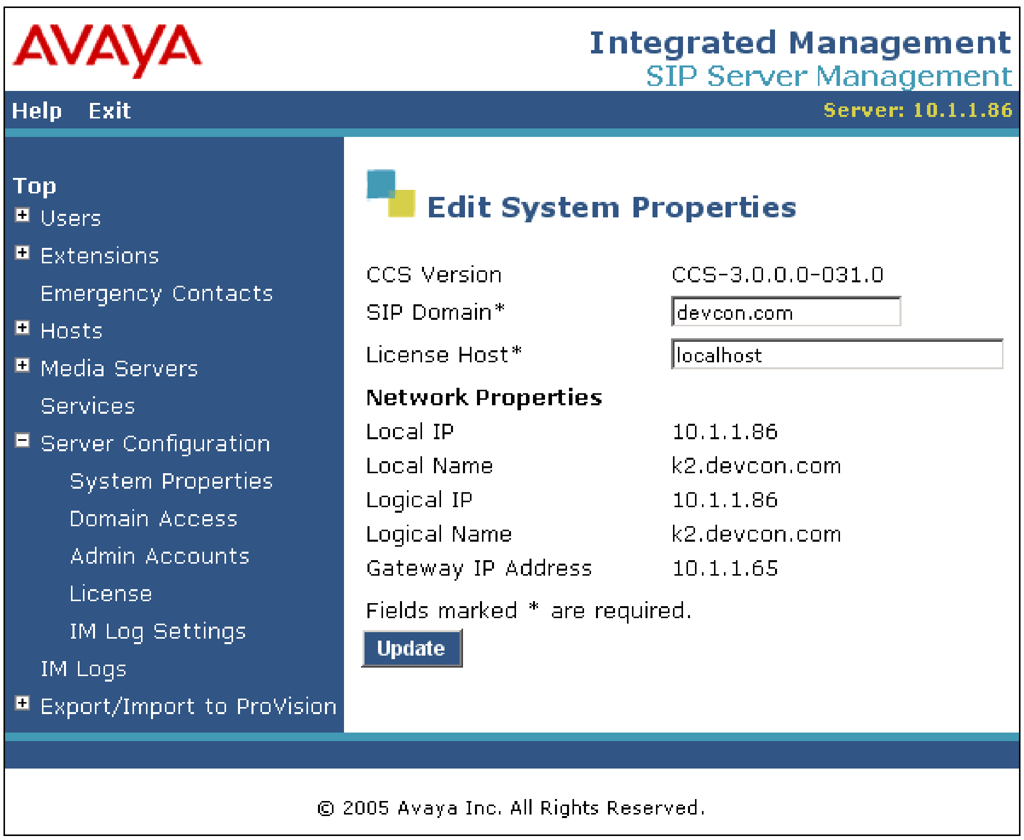

Figure 23: System Properties

Step 3: Enter Avaya SES Host Information
After setting up the domain in the **Edit 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.

The **Edit Host** screen shown in **Figure 24** 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 23**) of this server in the **Host IP Address** field.
- Enter the **DB Password** that was specified while running the install script during the system installation.
- The default values for the other fields may be used as shown in **Figure 24**.
- Click the **Update** button.

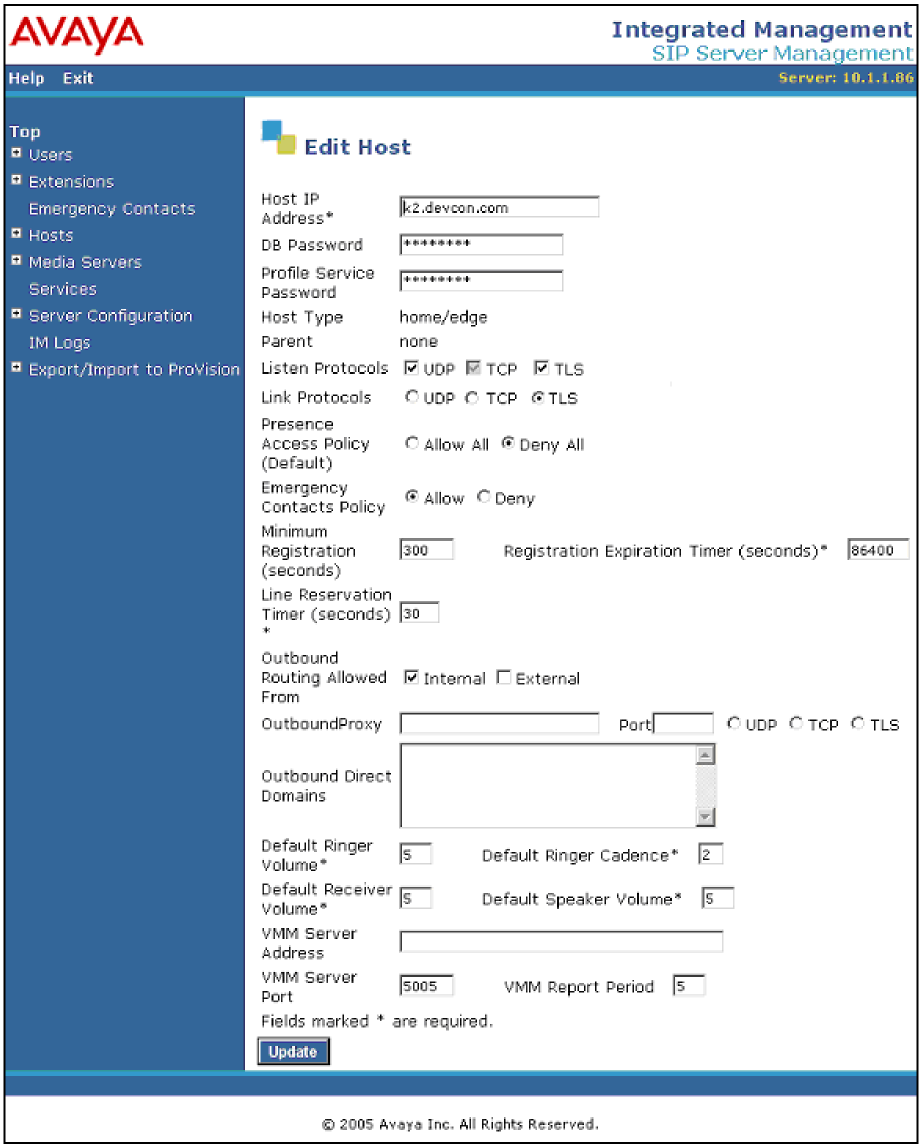

Figure 24: Edit Host

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., S8710-CLAN).
- Select the home SES server in the **Host** field as specified in **Figure 24**.
- 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 S8710 Media Server CLAN board in the **SIP Trunk IP Address** field. (Note: This may be the IP address of the media server processor in smaller Avaya Communication Manager configurations using an Avaya S8300 Media Server.)
- After completing the **Add Media Server** screen, click on the **Add** button.

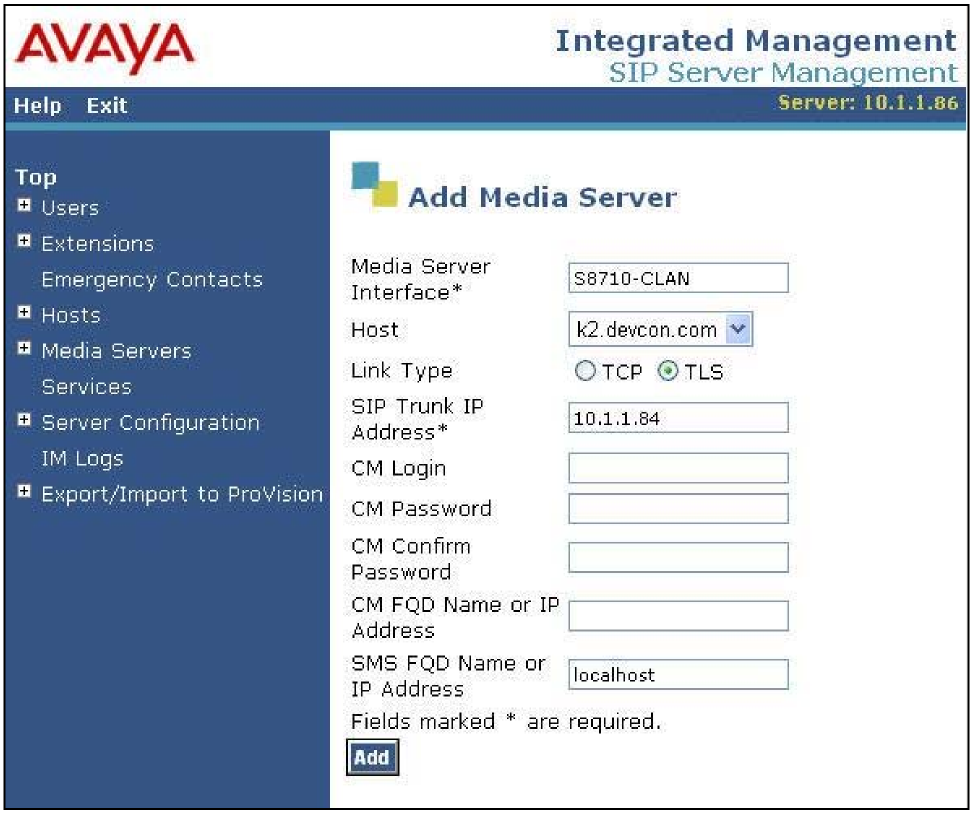

Figure 25: Add Media Server

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 Manager systems 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 PAETEC Communications IPATH Service, the *user* portion of the SIP URI will contain the 10 digit value specified for the incoming direct inward dialed telephone number.

An example of a SIP URI in an INVITE message received from PAETEC Communications would be:

sip:5853079961@10.1.1.86;user=phone;npdi=yes

The user portion in this case is the 10-digit DID number "5853079961".

The strategy used to define the media server address maps will be to create a pattern that matches the DID numbers assigned to the customer by PAETEC Communications. The SES will forward the messages with matching patterns to the appropriate CLAN interface of the S8710 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.
- 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 26** 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 these Application Notes, the DID numbers provided by PAETEC Communications are 585-307-9900 thru 9999. The pattern specification (without the double quotes) for DID numbers assigned is: `"^sip:58530799[0-9]{2}"`. This means that URIs beginning with "sip:58530799" followed by any other 2 digits will match the pattern and be routed to the interface defined for S8710-CLAN.

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

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

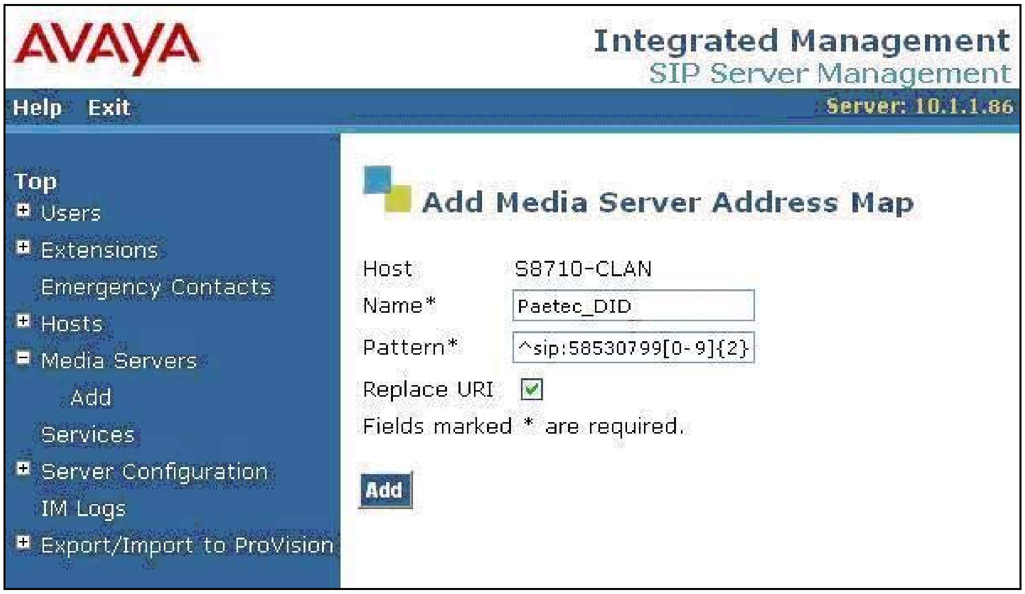

Figure 26: Media Server Address Map

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

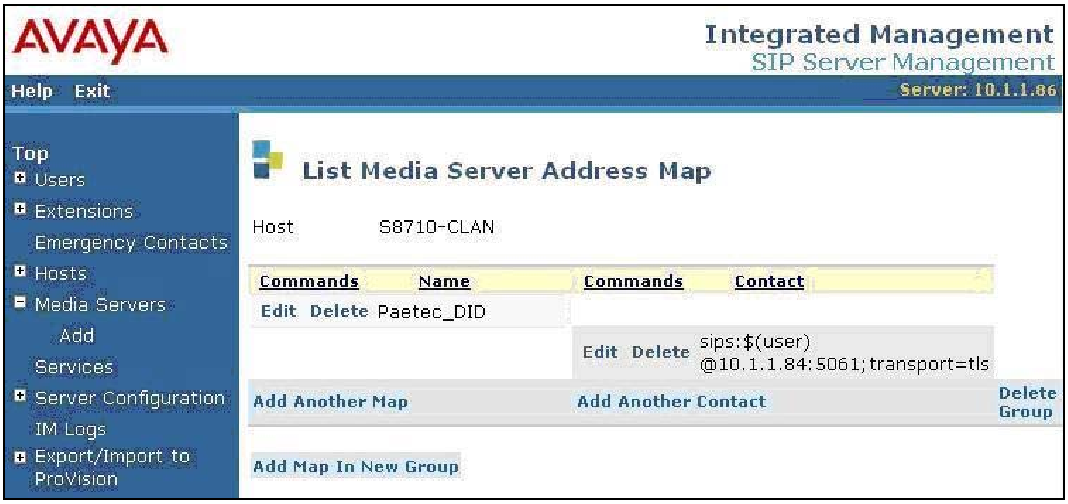

Figure 27: List Media Server Address Map

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 26**, the following contact was created:

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

The contact specifies the IP address of the CLAN and the transport protocol used to send SIP signaling messages. The incoming DID number sent in the user part of the original request URI is substituted for `$(user)`.

Step 6: Specify Address Maps to PAETEC Communications

Outbound PSTN calls are directed by the Avaya Communication Manager ARS feature according to the customer's network design guidelines. These guidelines determine what types of outgoing calls should be sent to the PAETEC Communications IPATH Service. The ARS routing decisions (for trunk group selection) will be customer specific and are beyond the scope of these notes.

SIP signaling messages for outbound calls sent to the SIP trunk are then routed to the PAETEC Communications gateway using Host Address Maps within Avaya SIP Enablement Services. As with the inbound media server address maps, these Host Address Maps use pattern matching on the SIP URI to direct messages to the corresponding contact address (e.g., the PAETEC Communications SIP signaling gateway).

In this configuration, the Avaya SES routing rule for the SIP trunk group will be to send all outbound PSTN traffic to the PAETEC Communications IPATH Service. To perform this, several dialing patterns will be created in the 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).
- Finally N11 service codes (such as 411, 611, etc.) will be recognized using the pattern "`^sip:[2-9]11`".

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

The configuration of the host address map for all 1 plus 10-digit North American calls is shown in **Figure 28**.

- 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 28**.
- Enter a descriptive name for the map, such as “PAETEC_1Plus10”.
- Specify an appropriate pattern for the call type. In this example, the pattern used for North American calls is “`^sip:1[0-9]{10}`”.
- Leave the **Replace URI** checkbox selected.
- Click the **Add** button.

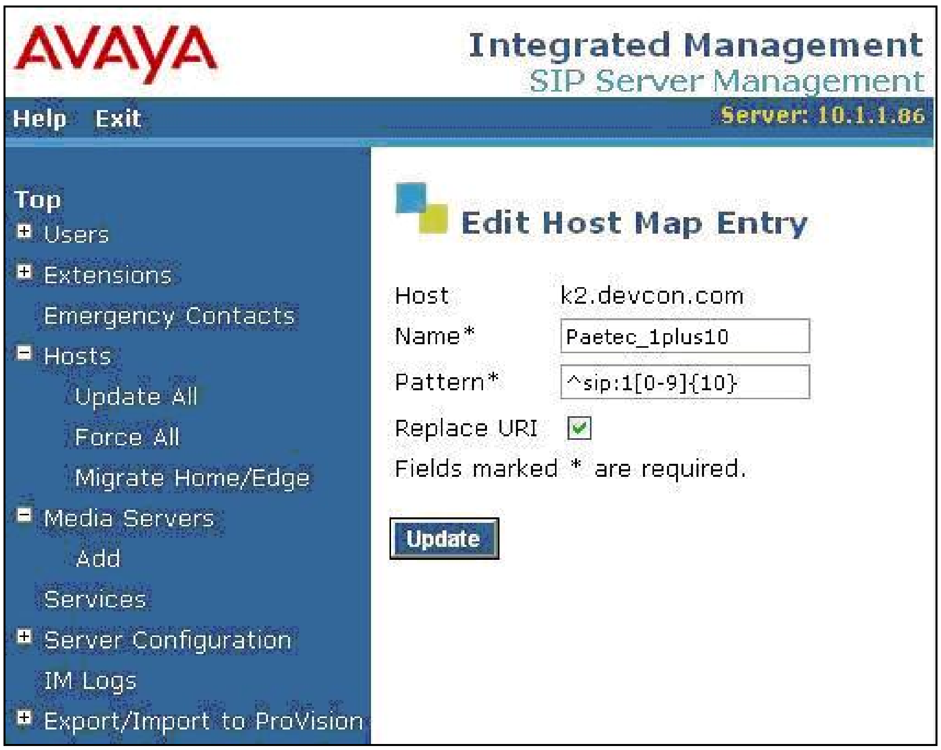

Figure 28: Edit Host Map Entry

Additional Host Address Map patterns are added in a similar manner. **Figure 29** illustrates the entry for Operator "zero" and "zero-plus" dialing.

Figure 29: Address Map for 0 and 0+ Dialing

Figure 30 illustrates the host address map for the N11 service codes.

Figure 30: N11 Host Address Map

Step 7: Specify the PAETEC Communications SIP Gateway Information
The next step is to enter the contact address for the PAETEC Communications SIP gateway. In this example, the IP address 20.1.1.135 is used. The customer must obtain the actual address of this gateway from PAETEC Communications.

To enter the PAETEC Communications SIP gateway 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 28** and **Figure 29** 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)@20.1.1.135:5060;transport=udp
  ```

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

After configuring the host address maps and contact information, the **List Host Address Map** screen will appear as shown in **Figure 31**.

Figure 31: 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 32**.

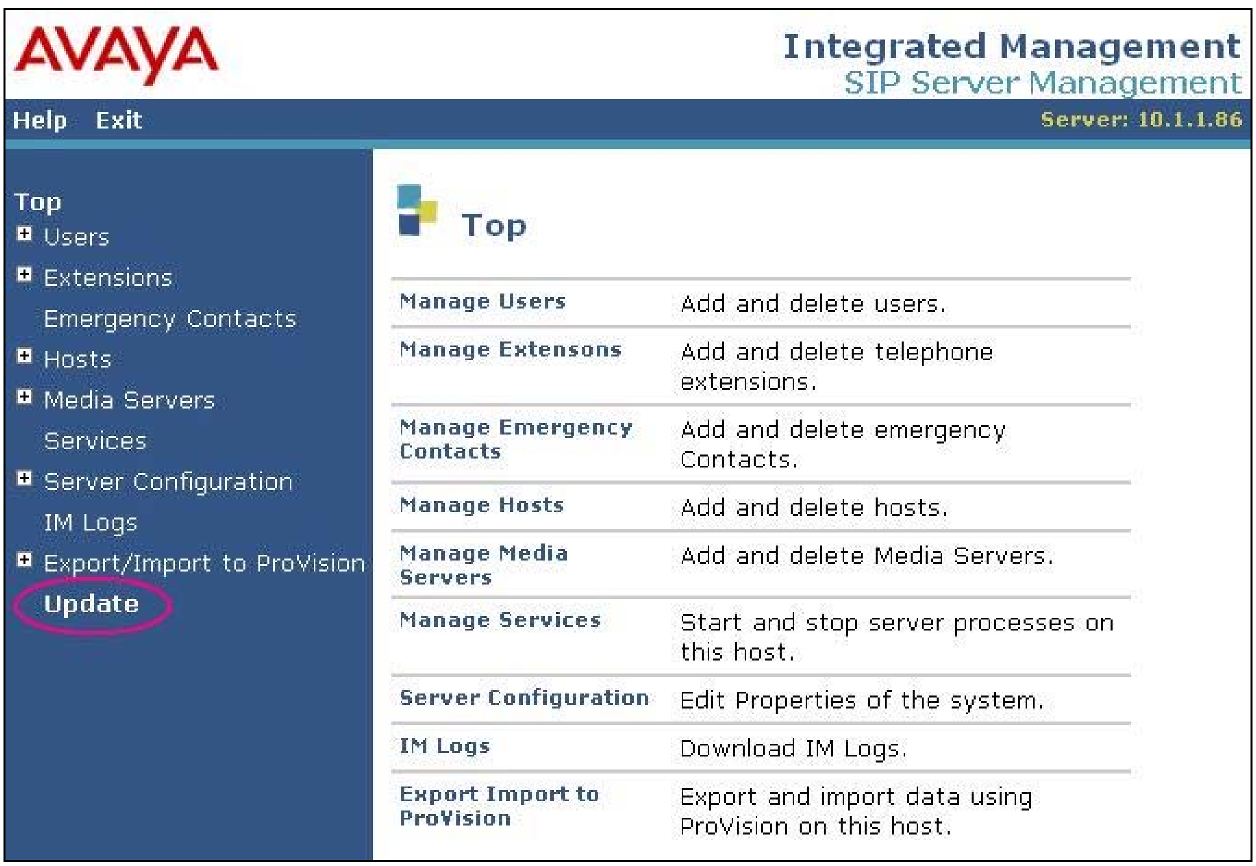

Figure 32: Update Following SES Administrative Changes

Step 9: Specify the PAETEC Communications SIP Gateway as a Trusted Host
The final step to complete the SIP trunk administration on Avaya SES is to designate the IP address of PAETEC Communications SIP Gateway as a trusted host. As a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the designated IP address.[1]

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:

- Log in to the Avaya SES using the administrative login and password.

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

  ```
  trustedhost –a 20.1.1.135 –n k2.devcon.com –c PAETEC_Gway
  ```

  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 33 illustrates the results of the trustedhost commands.[2]

- 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 32**.

  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.

---

[1] Note, if the trusted host step is not done, authentication challenges to incoming SIP messages (such as INVITEs and BYEs) will be issued by the SES. This may cause call setup to fail, active calls to be disconnected after timeout periods, and/or SIP protocol errors.
[2] For completeness, the –d argument allows the trust relationship to be deleted. For example,

```
trustedhost –d 20.1.1.135 –n k2.devcon.com
```

removes the trust relationship added above.

```
admin@k2> trustedhost -a 20.1.1.135 -n k2.devcon.com –c PAETEC_Gway
20.1.1.135 is added to trusted host list.

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

Figure 33: 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 basic instructions for completing the administration necessary to support the optional Avaya 4600 Series SIP telephones. Additional features, such as the use of mnemonic addressing, 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.

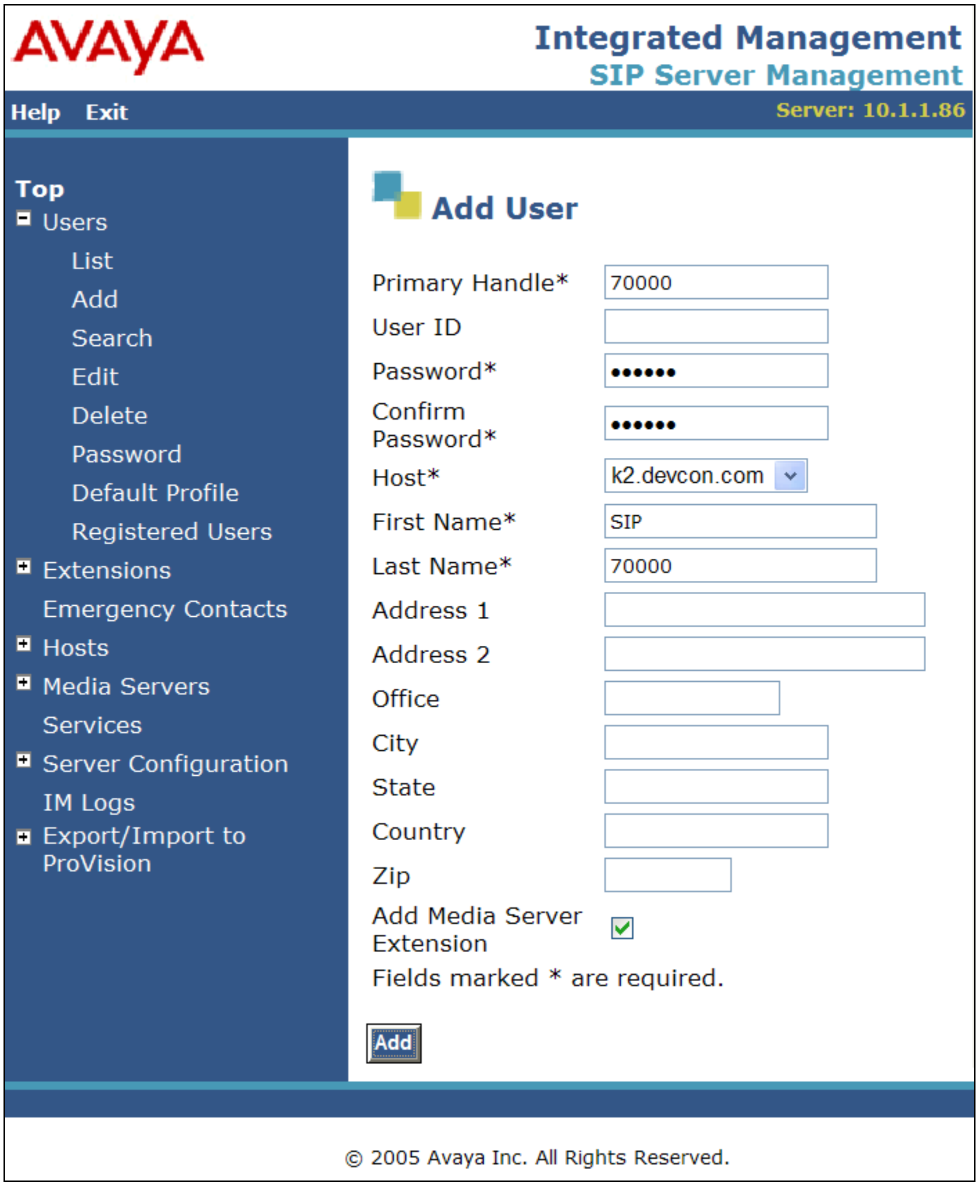

Figure 34: Add User

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 16**, 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.

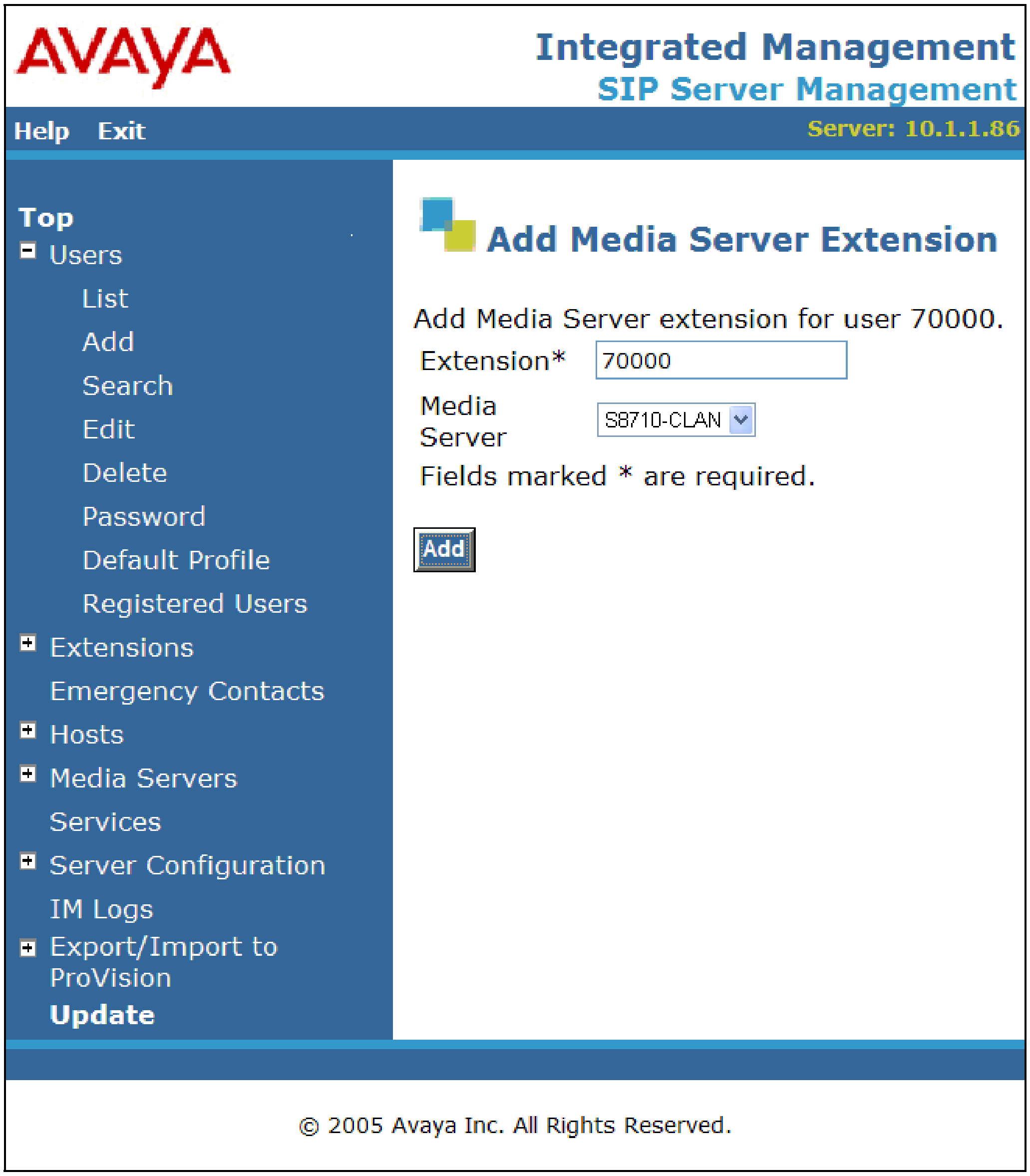

Figure 35: Add Media Server Extension

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

5. PAETEC Communications IPATH Service Configuration

In order to use PAETEC Communications IPATH Service, a customer must request service from PAETEC using their sales processes. The process can be started by contacting PAETEC Communications via their corporate web site at http://www.paetec.com/contact/inforequest.asp and requesting information via the online sales links or telephone numbers.

During the signup process, PAETEC 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).

For these Application Notes, PAETEC Communications provided the following information:

IP address of the PAETEC Communications SIP gateway	20.1.1.135
Direct Inward Dialed (DID) numbers	585-307-9900 through 9999
Codecs supported	G.711mu, G.729B

This information was used to complete the Avaya Communication Manager and Avaya SIP Enablement Services administration discussed in the previous sections.

6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the PAETEC Communications IPATH Service and an Avaya SIP Telephony Solution using SIP Trunking. This section covers the general test approach and the test results.

6.1. General Test Approach

A simulated enterprise site consisting of an Avaya SIP 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 IPATH SIP Trunking Service provided by PAETEC Communications. This allowed the enterprise site to use SIP trunking for PSTN 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 PAETEC Communications.
- Outgoing calls from the enterprise site were completed via PAETEC Communications to the PSTN destinations.
- Calls using SIP, H.323, digital and analog endpoints supported by the Avaya SIP Telephony solution.
- Various call types including: local, long distance, international, toll free, operator and directory assistance calls.
- Calls using G.711mu and G.729B codecs.
- DTMF transmission using RFC 2833.
- Telephone features such as hold, transfer and conference.
- 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 tests were completed successfully.

7. Verification Steps

This section provides the 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 calls through PAETEC Communications.

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-to-IP media 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 PAETEC Communications. 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 PAETEC Communications IPATH Service, contact PAETEC Communications Customer Service at 877-340-2600 or customerservice@PAETEC.com Include the customer account number in the communication.

9. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya Communication Manager and Avaya SIP Enablement Services telephony solution to the PAETEC Communications IPATH Service using SIP trunking. The PAETEC Communications IPATH Service is a Voice over IP solution for customers ranging from small businesses to large enterprises. SIP trunking uses the Session Initiation Protocol to connect private company networks to the public telephone network via converged IP access. It provides businesses a flexible, cost-saving alternative to traditional hardwired telephony trunk lines.

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*, May 2006, Issue 2.1, Document Number 03-300509.

[2] *Feature Description and Implementation for Avaya Communication Manager*, Issue 4, Document Number 555-245-205.

[3] *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.

[4] *Converged Communications Server R3.0 Installation and Administration Guide (SIP Enablement Services R3.0)*, July 2005, Issue 5.1, Document Number 555-245-705.

[5] *SIP Support in Release 3.1 of Avaya Communication Manager Running on the Avaya S8300, S8500, S8500B, S8700, and S8710 Media Server*, February 2006, Issue 6, Document Number 555-245-206.

[6] *4600 Series IP Telephone R2.4 LAN Administrator Guide*, April 2006, Document Number 555-233-507.

Additional information about PAETECH Communications IPATH Service is available at http://www.paetec.com.

APPENDIX A: Sample SIP INVITE Messages

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

```
INVITE sip:5853079961;npdi@10.1.1.86;user=phone SIP/2.0
Via: SIP/2.0/UDP 20.1.1.135:5060;branch=z9hG4bK0odoaj0070ag0akrt7o0.1
From: "AVAYA INC C/O
T"<sip:7328571637@20.1.1.135:5060;user=phone;avaya1plex=AVAYA1PLEX-
q2jm5jv0cfg77>;tag=10000000-0-1108989597
To: <sip:5853079961@10.1.1.86;user=phone>
CSeq: 1 INVITE
Contact: "AVAYA INC C/O T"<sip:7328571637@20.1.1.135:5060;avaya1plex=AVAYA1PLEX-
q2jm5jv0cfg77;transport=udp>
Call-ID: 10DA5B20-4E7B6@10.254.1.7
P-Asserted-Identity: "AVAYA INC C/O T"<sip:7328571637@10.254.1.7;user=phone>
Privacy: none
Max-Forwards: 69
Content-Type: application/sdp
Content-Length: 218

v=0
o=- 3355764663 3355764663 IN IP4 20.1.1.135
s=-
c=IN IP4 20.1.1.135
t=0 0
m=audio 20038 RTP/AVP 0 8 18 13
a=ptime:20
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:13 CN/8000
```

Sample SIP INVITE Message from Avaya SIP Enablement Services to PAETEC Communications:

```
INVITE sip:15853402870@20.1.1.135:5060;transport=udp SIP/2.0
Call-ID: 0e21859e20db1302444ce3fad00
CSeq: 1 INVITE
From: "Dcp Phone62004"
<sip:15853079963@east.devcon.com:5061>;tag=0e21859e20db12f2444ce3fad00
Record-Route: <sip:10.1.1.86:5060;lr>,<sip:10.1.1.84:5061;lr;transport=tls>
To: "15853402870" <sip:15853402870@east.devcon.com>
Via: SIP/2.0/UDP 10.1.1.86:5060;branch=z9hG4bK03030356565623232333f.0,SIP/2.0/TLS
10.1.1.84;psrrposn=2;branch=z9hG4bK0e21859e20db1312444ce3fad00
Content-Length: 201
Content-Type: application/sdp
Contact: "Dcp Phone62004" <sip:15853079963@10.1.1.84:5061;transport=tls>
Max-Forwards: 69
User-Agent: Avaya CM/R013x.01.0.628.6
Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
Session-Expires: 1800;refresher=uac
Min-SE: 1800
History-Info: <sip:15853402870@east.devcon.com>;index=1
History-Info: "15853402870" <sip:15853402870@east.devcon.com>;index=1.1
Supported: 100rel,timer,replaces,join,histinfo
P-Asserted-Identity:"Dcp Phone62004"<sip:15853079963@east.devcon.com:5061>

v=0
o=- 1 1 IN IP4 10.1.1.84
s=-
c=IN IP4 10.1.1.116
t=0 0
m=audio 2432 RTP/AVP 0 8 18 127
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:127 telephone-event/8000
```

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).
  - An asterisk '**`*`**' matches zero or more of the preceding characters.
  - Square brackets enclose a list of any character to be matched. Ranges are designated by using a hyphen. Thus the expression **`[12345]`** or **`[1-5]`** both describe a pattern that will match any single digit between 1 and 5.
  - Curly brackets containing an integer 'n' indicate that the preceding character must be matched exactly 'n' times. Thus **`5{3}`** matches '555' and **`[0-9]{10}`** indicates any 10 digit number.
  - The circumflex character **`^`** as the first character in the pattern indicates that the string must begin with the character following the circumflex.

Putting these constructs together as used in this document, the pattern to match the SIP INVITE string for any valid 1+ 10-digit number in the North American dial plan would be:

`^sip:1[0-9]{10}`

This reads as: "Strings that begin with exactly **sip:1** and having any 10 ddigits following will match.

A typical INVITE request below uses the shaded portion to illustrate the matching pattern.

```
INVITE sip:17325551638@20.1.1.54:5060;transport=udp SIP/2.0
```

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Developer*Connection* Program at devconnect@avaya.com.