Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking Using PAETEC Communications Dynamic IP SIP Trunk Service and an Avaya IP Telephony Solution – 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the PAETEC Communications Dynamic IP SIP Trunk Service and an Avaya IP telephony solution. PAETEC can offer Dynamic IP SIP Trunk Service using several different platform technologies in the network. These Application Notes correspond to Dynamic IP SIP Trunk Service offered using a Lucent platform in the network. The Avaya solution consists of Avaya SIP Enablement Services, Avaya Communication Manager, and various Avaya SIP, H.323, digital and analog endpoints.

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

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the PAETEC Dynamic IP SIP Trunk 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.

PAETEC can offer the Dynamic IP SIP Trunk Service using several different platform technologies in the PAETEC network. These Application Notes correspond to Dynamic IP SIP Trunk Service offered using a Lucent platform in the network. The PAETEC and Avaya platforms illustrated in these Application Notes had been previously compliance-tested using earlier versions of all components, as documented in Reference [1]. The compliance testing associated with these Application Notes updates reference [1] using current versions of the platforms.

Customers using this Avaya IP telephony solution with the PAETEC Dynamic IP SIP Trunk Service are able to place and receive PSTN calls via a dedicated broadband Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

The text and coverage diagram below summarizes the PAETEC Dynamic IP SIP Trunk Service at the time of writing these Application Notes. Please consult PAETEC for the most current description of capabilities. PAETEC serves 82 of the top 100 Metropolitan Statistical Areas, and offers data, voice, and value-added services throughout the United States. From local and long distance to VoIP, PAETEC offers a full spectrum of traditional and next-generation voice services, each predicated on vast industry expertise and the world-class technology of partners.

PAETEC Dynamic IP SIP Trunk Service includes 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

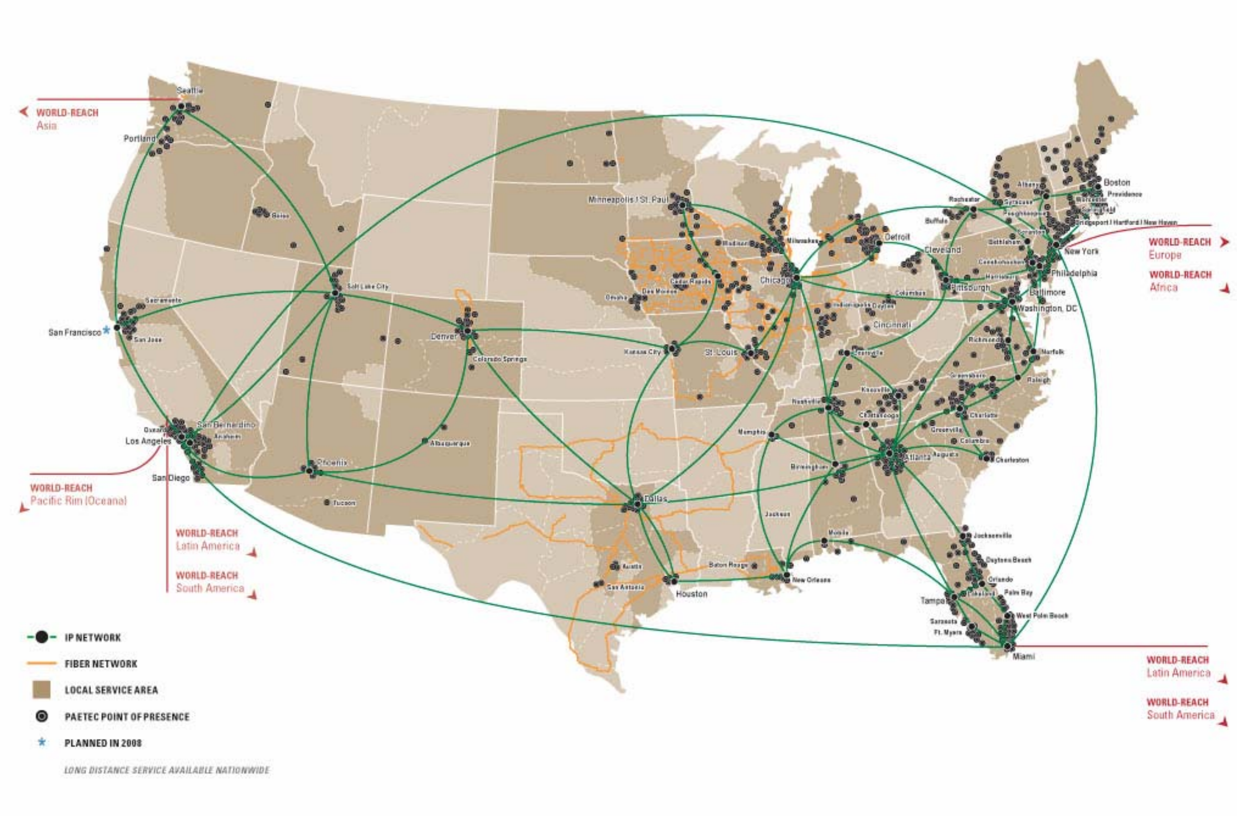

Figure 1 illustrates an example Avaya IP telephony solution connected to the PAETEC Dynamic IP SIP Trunk Service. This is the configuration used during the DevConnect compliance testing process. Please refer to **Section 6** for the features tested with this solution.

The Avaya components used to create a simulated customer site included:

- Avaya S8720 Servers running Avaya Communication Manager Release 5.1
- Avaya G650 Media Gateway and associated hardware
- Avaya SIP Enablement Services (SES) Release 5.1 on an Avaya S8500 Server platform
- Avaya 9600-Series IP telephones (configured for the SIP protocol)
- Avaya 9600-Series IP telephones (configured for the H.323 protocol)
- Avaya 4600-Series IP telephones (configured for the SIP protocol)
- Avaya 4600-Series IP telephones (configured for the H.323 protocol)
- Avaya digital phones
- Analog phones and fax machines
- Avaya IP Softphone

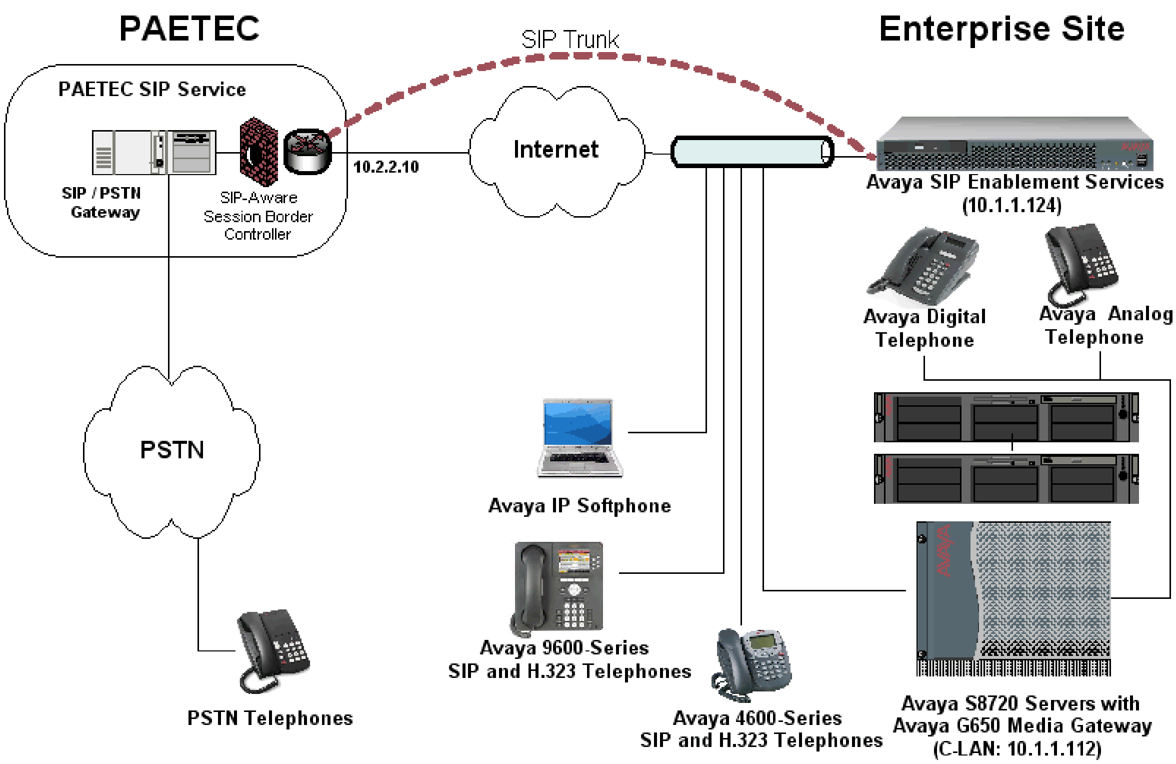

Figure 1: Avaya IP Telephony Network using PAETEC Dynamic IP SIP Trunk Service

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 an incoming PSTN call to the enterprise site. The call can terminate to an analog, digital, H.323, or SIP telephone at the enterprise site, as described below.

1. A user on the PSTN dials a PAETEC-provided DID number assigned to an Avaya Communication Manager telephone at the enterprise site. The PSTN routes the call to the PAETEC network. PAETEC then routes the DID number to the assigned customer.

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

3. Avaya SES routes the call to Avaya Communication Manager, also using a SIP trunk.

4. Avaya Communication Manager rings the analog, digital, or H.323 telephone, as shown in step 4.

- or –

4a. If the inbound call is to a SIP extension at the enterprise, Avaya Communication Manager transmits the appropriate SIP signaling via Avaya SES to the SIP telephone, as shown by the 4a arrow.

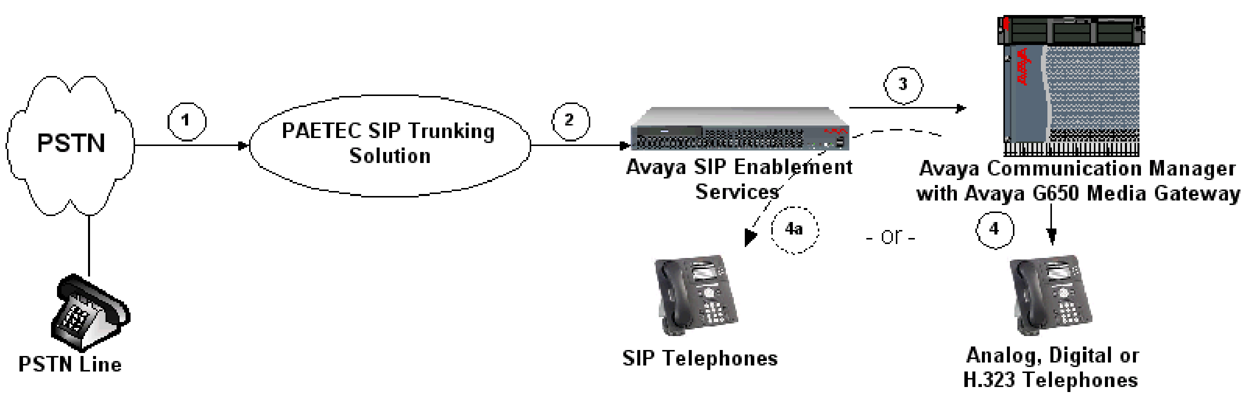

Figure 2: Incoming PSTN Calls to Avaya Communication Manager

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

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

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

- or-

1a. A 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 services and call routing are 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.

4. PAETEC completes the call to the PSTN.

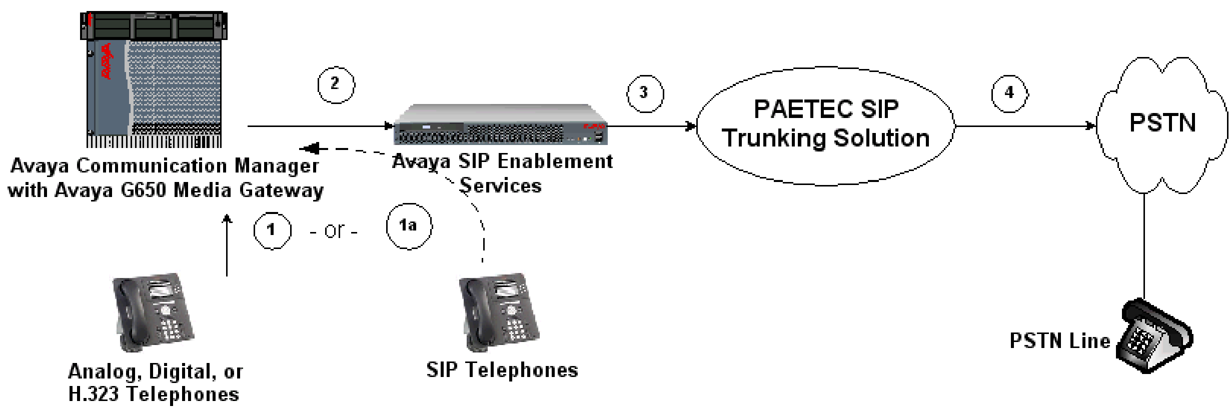

Figure 3: Outgoing Calls from Avaya Communication Manager to the PSTN

2. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Avaya S8720 Server with an Avaya G650 Media Gateway	Avaya Communication Manager Release 5.1 Load 414.3 Update: Service Pack 1 (15962)
Avaya SIP Enablement Services on S8500 Server	SES 5.1 Load 414.3f
Avaya 9640 IP Telephone	R2.0 – H.323
Avaya 9620 IP Telephone	R2.0.4 – SIP
Avaya 4610SW IP Telephone	R2.2.2 – SIP
Avaya 4621SW IP Telephone	R2.9 – H.323
Avaya IP Softphone	Release 6.0
Avaya 6416 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
PAETEC Dynamic IP SIP Trunk Service Solution Components	
Lucent Distributed Network Controller/Gateway	6.3.1.2SP3
Acme Packet Session Border Controller	2.01P64

Table 1: Equipment and Software Tested

The specific configuration above was used for the compatibility testing. Note that this solution will be compatible with other Avaya 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 Avaya Communication Manager for SIP Trunking. SIP trunks are established between Avaya Communication Manager and Avaya SIP Enablement Services (SES). These SIP trunks will carry SIP signaling associated with the PAETEC Dynamic IP SIP Trunk Service as well as signaling associated with SIP endpoint devices.

Avaya SIP telephones are configured as off-pbx stations (OPS) on Avaya Communication Manager. These 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.

The use of SIP endpoints is optional. The steps discussed in Sections 3.2 and 4.3 describing SIP endpoint 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 the PAETEC Dynamic IP SIP Trunk Service. There is no direct SIP signaling path between PAETEC and Avaya Communication Manager or Avaya SIP endpoints.

For incoming calls, the Avaya SES uses address maps to direct the incoming SIP messages to the appropriate Avaya Communication Manager, as shown in Section 4.1. Once the message arrives at Avaya Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions 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. The Avaya SES directs the outbound SIP messages to the PAETEC network.

The dial plan for the configuration described in these Application Notes consists of 1+10-digit dialing for local and long-distance calls over the PSTN. In addition, Directory Assistance calls (411) and International calls (011+Country Code) were also supported. Avaya Communication Manager routes all calls to the PAETEC network using Automatic Route Selection (ARS).

Avaya Communication Manager configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. Note that the IP Addresses shown throughout these Application Notes have been edited so that the actual IP Addresses of the network elements are not revealed. The general installation of the Avaya S8720 Server, Avaya G650 Media Gateway and circuit packs such as the C-LAN is presumed to have been previously completed and is not discussed here.

3.1 SIP Trunk Configuration

Step 1: Confirm Necessary Optional Features

Log into the Avaya Communication Manager SAT interface and confirm sufficient unused SIP trunk and Off-PBX Telephone capacities. Use the **display system-parameters customer-options** command to determine these values as shown in **Figure 4**. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

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

     G3 Version: V15                              Software Package: Standard
       Location: 1                             RFA System ID (SID): 1
       Platform: 6                             RFA Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 44000 62
                                      Maximum Stations: 36000 7
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 100   2
                    Maximum Off-PBX Telephones -   OPS: 100   2
```

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

On Page 2, verify that the **Maximum Administered SIP Trunks** supported by the system is sufficient for the combination of trunks to the PAETEC network, SIP endpoints and any other SIP trunks used. Each Avaya SIP telephone on a 2-party call with PAETEC uses two SIP trunks for the duration of the call. Each non-SIP telephone (i.e., analog, digital, H.323) on a 2-party call with PAETEC uses one SIP trunk.

```
display system-parameters customer-options                      Page   2 of  10
                                OPTIONAL FEATURES
IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 2000  0
           Maximum Concurrently Registered IP Stations: 12000 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: 2000  32
```

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

Subsequent pages of the form shown above can reveal whether other commonly used features, such as ARS and IP Stations, are enabled by the license file.

Step 2: Assign Node Names

The node names defined here will be used in other configuration screens to define a SIP signaling group between Avaya Communication Manager and Avaya SES. In the **IP Node Names** form, assign the node name and IP address for the Avaya SIP Enablement Services Server (SES) at the enterprise site as shown in **Figure 6**. In this case, “SES” and “10.1.1.124” are the name and IP Address for the Avaya SES, and “06A_CLAN” and “10.1.1.112” are the name and IP address assigned to a TN799DP C-LAN card to be used for SIP signaling. The C-LAN was previously created during the installation of the system. In other Avaya configurations such as an Avaya G250, G350, G700, or G450 Media Gateway with a standalone Avaya S8300 Server, the Avaya S8300 Server processor address (node name “procr”) must be used as the SIP signaling interface to Avaya SES, rather than a C-LAN interface.

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
06A_CLAN            10.1.1.112
10A_CLAN            192.168.100.18
11A_medpro          192.168.100.19
SES                 10.1.1.124
```

Figure 6: IP Nodes Names Form

Step 3: Define IP Network Regions

In the sample configuration used for compliance-testing, two network regions are used. Network region 1, the default region, is used for Avaya devices. The PAETEC Dynamic IP SIP Trunk Service will be logically defined as network region 2. Although thorough coverage of network regions is beyond the scope of these Application Notes, a brief summary follows. Analog and digital devices can derive a network region from the configuration of the gateway or cabinet to which the device is connected. Avaya IP Telephones, both H.323 and SIP, can derive a network region from an IP network map, that associates ranges of IP addresses with a network region. In the absence of a defined IP network mapping, an Avaya H.323 IP Telephone will be considered to be in the network region of the C-LAN to which it has registered, and an Avaya SIP Telephone will be considered to be in the network region defined for its associated SIP signaling group. Other devices, such as C-LANs, Media Processors, and Media Gateways can be specifically configured to a network region.

By using unique network regions for sets of devices or networks, finer control over behaviors such as codec selection and quality of service markings are possible. For example, one codec set may be used for intra-region connections among local Avaya devices, optimizing for quality using an uncompressed codec over a switched LAN. Another codec set may be used for inter-region connections between local Avaya devices and the PAETEC network components, perhaps optimizing for bandwidth conservation using a compressed codec, if WAN bandwidth is at a premium. This approach is illustrated in the screens in these Application Notes, where G.729A is used over the WAN to PAETEC, and G.711MU is used for local intra-region connections. During compliance testing, variations of the illustrated configuration were also tested, including G.711A, G.711MU, and G.729A for the connections to the PAETEC network.

Use the **change ip-network-region 1** command to set the following values:

- The **Authoritative Domain** field is configured to match the domain name configured on the Avaya SES. In this configuration, the domain name is "sipsp.avaya.com".
- By default, **IP-IP Direct Audio** (both **Intra-** and **Inter-Region**) is enabled to allow audio traffic to be sent directly between endpoints without using gateway resources such as the TN2602AP IP Media Resource card. PAETEC supports "shuffling" to direct **IP-IP Direct Audio** so these parameters can retain the "enabled" default values.
- The **Codec Set** on page 1 is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set 1 will be used for intra-region communication among the Avaya devices.

Although not highlighted, note also that the **IP Network Region** form is used to set the QoS packet parameters that provides priority treatment for signaling and audio packets over other data traffic. These parameters may need to be aligned with the specific values expected by PAETEC.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: sipsp.avaya.com
    Name: Avaya devices
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 60001
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5
```

Figure 7: IP Network Region 1 – Page 1

Navigate to page 3. In the bold row defining the communication between network region 1 and network region 2, set the **codec set** column to 2 as shown below. In the sample configuration, codec set 2 will therefore be used for connections between Avaya devices and the PAETEC network, which will logically reside in network region 2.

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

                   Inter Network Region Connection Management

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

Figure 8: IP Network Region 1 – Page 3

Use the **change ip-network-region 2** command to set the following values:

- By default, **IP-IP Direct Audio** (both **Intra-** and **Inter-Region**) are enabled to allow audio traffic to be sent directly between endpoints without using gateway resources such as the TN2602AP IP Media Resource card.
- The **Codec Set** on page 1 is set to the number of the IP codec set to be used for calls within IP network region 2. In this case, codec set 2 will be used for intra-region communication among the PAETEC SIP trunks, which in general is possible for cases such as off-net call forwarding or trunk-trunk transfer, where a call that came in on the SIP Trunk from PAETEC also goes out the SIP trunk to PAETEC.

```
change ip-network-region 2                                          Page   1 of  19
                               IP NETWORK REGION
  Region: 2
Location:         Authoritative Domain:
    Name: PAETEC Trunk
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5
```

Figure 9: IP Network Region 2 – Page 1

Navigate to page 3. In the bold row defining the communication between network region 2 and network region 1, observe that the **codec set** column is already set to 2, due to the previous configuration of network region 1. In the sample configuration, codec set 2 will be used for connections between Avaya devices and the PAETEC network.

```
 change ip-network-region 2                                         Page   3 of  19
                   Inter Network Region Connection Management

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

Figure 10: IP Network Region 1 – Page 3

Step 4: Define IP Codecs

Open the **IP Codec Set** form used for intra-region connections among the local Avaya devices using the codec specified in the **IP Network Region** form (**Figure 7**). Enter the list of audio codecs eligible to be used for local connections, in order of preference. The settings of the **IP Codec Set** form are shown in **Figure 11.** 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. As discussed in Step 3, G.711MU will be configured as the preferred codec for local connections. The inclusion of G.729A as a second choice in codec set 1 allows calls using Avaya 9600-Series SIP telephones to shuffle to ip-direct media using G.729A for calls to and from the PAETEC network. During compliance testing, other codec set configurations were also verified.

```
change ip-codec-set 1                                          Page   1 of   2
                          IP Codec Set
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2: G.729A            n         2        20
 3:
 4:
 5:
 6:
 7:
```

Figure 11: IP Codec Set 1 Form

Open the **IP Codec Set** form used for connections between network region 1 and 2 using the codec specified in page 3 of the **IP Network Region** form (**Figure 8**). Enter the list of audio codecs eligible to be used for connections to the PAETEC network, in order of preference. The settings of the **IP Codec Set** form are shown in **Figure 12.** As discussed in Step 3, G.729A is the codec to be used for connections to the PAETEC network. During compliance testing, other codec set 2 configurations were also verified, including voice over G.711MU and G.711A, and fax over G.711MU. See additional notes below regarding fax.

```
change ip-codec-set 2                                          Page   1 of   2
                          IP Codec Set
    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.729A            n         2        20
 2:
 3:
 4:
 5:
 6:
 7:
```

Figure 12: IP Codec Set 2 Form – Page 1

The PAETEC network does not support the T.38 fax protocol. If calls involving fax machines will be made using the PAETEC network, it is necessary to disable fax relay protocols by setting the **Fax Mode** to “off” on page 2 of the codec set form as shown below. If fax is used, an uncompressed codec such as G.711MU protocol would also need to be specified on page 1 of the codec set used for the fax call.

```
change ip-codec-set 2                                          Page   2 of   2
                          IP Codec Set

                              Allow Direct-IP Multimedia? n
                    Mode                 Redundancy
    FAX             off                   0
    Modem           off                   0
    TDD/TTY         US                    3
    Clear-channel   n                     0
```

Figure 13: IP Codec Set 2 Form – Page 2

Step 5: Configure the Signaling Groups

Three SIP signaling groups are configured. One "PSTN Outbound" signaling group (and trunk group) will be used for outbound PSTN calls to the PAETEC network. Another "PSTN Inbound" signaling group (and trunk group) will be used for inbound calls from the PAETEC network. A third "SIP OPS" signaling group is defined for calls involving SIP telephones. Recall that SIP telephones register with the Avaya SES and leverage the calling privileges and features provided by Avaya Communication Manager. The configuration steps below show the configuration of these signaling groups.

Configure the PSTN Outbound **Signaling Group** using the **add signaling group 2** command shown in **Figure 14** as follows:

- Set the **Group Type** field to *sip*
- The **Transport Method** field will default to *tls* (Transport Layer Security).
- Set the **Near-end Node Name** to an Avaya C-LAN card (node name "06A_CLAN"). This value is taken from the **IP Node Names** form shown in **Figure 6**.
- Set the **Far-end Node Name** to the node name defined for the Avaya SIP Enablement Services Server (node name "SES"), also shown in **Figure 6**.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region value used for PAETEC network, as shown in **Figure 9.** This field logically establishes the "far-end" for calls using this signaling group as network region 2. For calls from Avaya devices, the "near-end" will be network region 1. Therefore, connections between Avaya devices and the PAETEC network will be between region 1 and region 2.
- Enter the IP Address of the PAETEC network element (provided by PAETEC) in the **Far-end Domain** field. (Recall that the IP Addresses shown in the screens in these Application Notes are not the actual IP Addresses used for compliance-testing). For outbound PSTN calls to PAETEC, this field sets the domain in the Uniform Resource Identifier (URI) of the SIP "To" address in the outbound INVITE message.
- The **Direct IP-IP Audio Connections** field is set to 'y'. PAETEC supports the Avaya **Direct IP-IP Audio** feature. This feature can be disabled if desired.
- 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, as specified in reference [11].

- The default values for the other fields may be used.

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

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

   Near-end Node Name: 06A_CLAN              Far-end Node Name: SES
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 2
      Far-end Domain: 10.2.2.10

                                             Bypass If IP Threshold Exceeded? n

        DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? n
        Enable Layer 3 Test? n
 Session Establishment Timer(min): 3              Alternate Route Timer(sec): 6
```

Figure 14: PSTN-Outbound Signaling Group Form

Configure the PSTN Inbound **Signaling Group** using the **add signaling group 3** command shown in **Figure 15** as follows:

- Set the **Group Type** field to *sip*
- The **Transport Method** field will default to *tls* (Transport Layer Security).
- Set the **Near-end Node Name** to an Avaya C-LAN card (node name “06A_CLAN”). This value is taken from the **IP Node Names** form shown in **Figure 6**.
- Set the **Far-end Node Name** to the node name defined for the Avaya SIP Enablement Services Server (node name “SES”), also shown in **Figure 6**.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region value used for PAETEC network, as shown in **Figure 9.** This field logically establishes the “far-end” for calls using this signaling group as network region 2. For calls to Avaya devices, the “near-end” will be network region 1. Therefore, connections from the PAETEC network to Avaya devices will be between region 2 and region 1.
- Leave the **Far-end Domain** field blank, allowing inbound PSTN calls from PAETEC to be accepted using this signaling group.
- The **Direct IP-IP Audio Connections** field is set to ‘y’. PAETEC supports the Avaya **Direct IP-IP Audio** feature. This feature can be disabled if desired.
- 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, as specified in reference [11].
- The default values for the other fields may be used.

```
add signaling-group 3                                           Page   1 of   1
                                 SIGNALING GROUP

 Group Number: 3              Group Type: sip
                        Transport Method: tls

   Near-end Node Name: 06A_CLAN             Far-end Node Name: SES
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 2
       Far-end Domain:

                                            Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
                                                       IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 3             Alternate Route Timer(sec): 6
```

Figure 15: PSTN-Inbound Signaling Group Form

Configure the SIP OPS **Signaling Group** using the **add signaling group 1** command shown in **Figure 16** as follows:

- Set the **Group Type** field to *sip*
- The **Transport Method** field will default to *tls* (Transport Layer Security).
- Set the **Near-end Node Name** to an Avaya C-LAN card (node name "06A_CLAN"). This value is taken from the **IP Node Names** form shown in **Figure 6**.
- Set the **Far-end Node Name** to the node name defined for the Avaya SIP Enablement Services Server (node name "SES"), also shown in **Figure 6**.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region value used for the local Avaya SIP Telephones**.** This field logically establishes the "far-end" for calls using this signaling group as network region 1.
- In the **Far-end Domain** field, enter the domain matching the domain specified on the Avaya SES and the Avaya local network region(s) (as shown in **Figure 7**).
- The **Direct IP-IP Audio Connections** field is set to 'y'.
- The **DTMF over IP** field should remain set to the default value of *rtp-payload*.
- 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: 06A_CLAN              Far-end Node Name: SES
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
       Far-end Domain: sipsp.avaya.com

                                             Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? n
         Enable Layer 3 Test? y
 Session Establishment Timer(min): 3              Alternate Route Timer(sec): 6
```

Figure 16: SIP OPS Signaling Group Form

Step 6: Configure the Trunk Groups
One trunk group will be associated with each of the signaling groups described in Step 5.

Configure the PSTN Outbound **Trunk Group** form as shown in **Figure 17** using the **add trunk-group** command. In this case, the trunk group number chosen is 2. On Page 1 of this form:

- Set the **Group Type** field to *sip*.
- Choose a descriptive **Group Name**.
- Specify a trunk access code (**TAC**) consistent with the dial plan
- Set the **Service Type** field to *public-ntwrk*.
- Specify the PSTN Outbound signaling group associated with this trunk group in the **Signaling Group** field as previously configured in **Figure 14**.
- Specify the **Number of Members** supported by this SIP trunk group.

One trunk member from this trunk group will be used for each outbound trunk call to the PAETEC network.

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

Group Number: 2                   Group Type: sip          CDR Reports: y
  Group Name: PAETEC-out                 COR: 1        TN: 1        TAC: 1002
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

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

Figure 17: Outbound PSTN Trunk Group Form – Page 1

Navigate to page 2 of the **Trunk Group** form. As shown in **Figure 18**, set the **Preferred Minimum Session Refresh Interval (sec)** field to at least “900”. If the default value of 600 is retained in this field, each outbound SIP call to PAETEC will require additional, avoidable SIP messaging that can perceptibly delay call establishment. With this value set to 900, the initial SIP INVITE message from Avaya to PAETEC will contain a value the PAETEC network finds acceptable, obviating the need for extra SIP messaging to establish mutually-acceptable session expiration and refresh timing for each call.

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

TRUNK PARAMETERS
     Unicode Name? y
                                            Redirect On OPTIM Failure: 5000

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

Figure 18: Outbound PSTN Trunk Group Form – Page 2

Navigate to page 3 of the **Trunk Group** form. As shown in **Figure 19**, set the **Numbering Format** field to "public".

```
add trunk-group 2                                                Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y
                     Numbering Format: public
                                               UUI Treatment: service-provider
                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n
```

Figure 19: Outbound PSTN Trunk Group Form – Page 3

Navigate to page 4 of the **Trunk Group** form. As shown in **Figure 20**, set the **Telephone Event Payload Type** (associated with DTMF transmission using RFC 2833) to the value "101". During compliance testing, the default value of "blank" was also verified successfully, suggesting that the PAETEC platforms used for the compliance test are capable of negotiating to an alternate telephone event payload type offered by Avaya. Nevertheless, PAETEC recommends the value "101" for their service.

```
add trunk-group 2                                              Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
             Telephone Event Payload Type: 101
```

Figure 20: Outbound PSTN Trunk Group Form – Page 4

Configure the PSTN Inbound **Trunk Group** form as shown in **Figure 21** using the **add trunk-group** command. In this case, the trunk group number chosen is 3. On page 1 of this form:

- Set the **Group Type** field to *sip*.
- Choose a descriptive **Group Name**.

- Specify a trunk access code (**TAC**) consistent with the dial plan
- Set the **Service Type** field to *public-ntwrk*.
- Specify the PSTN Inbound signaling group associated with this trunk group in the **Signaling Group** field as previously configured in **Figure 15**.
- Specify the **Number of Members** supported by this SIP trunk group.

One trunk member from this trunk group will be used for each inbound trunk call from the PAETEC network.

```
change trunk-group 3                                                Page   1 of  21
                                TRUNK GROUP
Group Number: 3                    Group Type: sip           CDR Reports: y
  Group Name: PAETEC Inbound              COR: 1        TN: 1        TAC: 1003
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                                       Signaling Group: 3
                                                     Number of Members: 10
```

Figure 21: Inbound PSTN Trunk Group Form – Page 1

Navigate to page 2 of the **Trunk Group** form. As shown in **Figure 22**, set the **Preferred Minimum Session Refresh Interval (sec)** field to at least “900”. If the default value of 600 is retained in this field, inbound SIP calls from PAETEC may incur the same type of avoidable SIP messaging described in the text above **Figure 18**. In this case, the avoidable extra messaging would be due to timer settings in Avaya SIP INVITE messages associated with “shuffling” procedures to ip-direct media, for incoming PAETEC trunk calls to IP Telephones.

```
add trunk-group 3                                                     Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS
     Unicode Name? y
                                            Redirect On OPTIM Failure: 5000
           SCCAN? n                                Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 900
```

Figure 23: Inbound PSTN Trunk Group Form – Page 2

Navigate to page 3 of the **Trunk Group** form. As shown in **Figure 24**, set the **Numbering Format** field to “public”. Since this trunk group is used for incoming PSTN trunk calls, optionally, the Avaya Communication Manager ability to replace restricted and unavailable numbers with a configurable text string can also be utilized, by enabling the fields shown in bold. The system-wide text string to appear on the display of a display-equipped telephone when an incoming call has caller id marked for privacy or has no caller id display info available can be configured on page 9 of the “system-parameters features” form (not shown). In the compliance-testing, the configurable replacement string for unavailable calls was observed on the display of Avaya telephones when a PSTN user requested restriction of the display of calling party information and called one of the PAETEC-provided DID numbers.

```
add trunk-group 3                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y
                     Numbering Format: public
                                                UUI Treatment: service-provider
                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y
```

Figure 24: Inbound PSTN Trunk Group Form – Page 3

Navigate to page 4 of the **Trunk Group** form. As shown in **Figure 25**, set the **Telephone Event Payload Type** to the value "101". During compliance testing, the default value of "blank" was also verified successfully. PAETEC recommends the value "101" for their service.

```
add trunk-group 3                                              Page   4 of  21
                              PROTOCOL VARIATIONS
                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
             Telephone Event Payload Type: 101
```

Figure 25: Inbound PSTN Trunk Group Form – Page 4

Configure the SIP OPS **Trunk Group** form as shown in **Figure 26** 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 descriptive **Group Name**.
- Specify a trunk access code (**TAC**) consistent with the dial plan
- Set the **Service Type** field to "tie".
- Specify the SIP OPS signaling group associated with this trunk group in the **Signaling Group** field, as configured in **Figure 16**.
- Specify the **Number of Members** supported by this SIP trunk group.

One trunk member from this trunk group will be used for each leg of a call to or from an Avaya SIP Telephone registered with the Avaya SES. For example, an outbound call from a SIP Telephone to PAETEC will use one trunk member from trunk group 1 and one trunk member from trunk group 2. An incoming call from PAETEC to a SIP Telephone will use one trunk member from trunk group 1 and one trunk member from trunk group 3.

```
add trunk-group 1                                                Page   1 of  21
                                TRUNK GROUP
Group Number: 1                    Group Type: sip           CDR Reports: y
  Group Name: SIP OPS to SES             COR: 1        TN: 1        TAC: 1001
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                                      Signaling Group: 1
                                                    Number of Members: 20
```

Figure 26: SIP OPS PSTN Trunk Group Form – Page 1

Navigate to page 3 of the **Trunk Group** form. As shown in **Figure 27**, set the **Numbering Format** field to "public".

```
change trunk-group 1                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
```

Figure 27: SIP OPS Trunk Group Form – Page 3

Step 7: Configure Calling Party Number Information

Use the **change public-unknown-numbering** command shown in **Figure 28** to configure Avaya Communication Manager to send the calling party number. In the sample configuration, all stations with a 5-digit extension beginning with 2 will send the calling party number 732-852-xxxx to PAETEC. This calling party number will be sent in the SIP "From" header, and displayed on display-equipped PSTN telephones.

```
change public-unknown-numbering 0                             Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext             Trk       CPN           CPN
Len Code            Grp(s)    Prefix        Len
                                                     Total Administered: 1
 5  2                         73285         10          Maximum Entries: 9999
```

Figure 28: Format For Calling Party Number

Step 8: Automatic Route Selection for Outbound Calls

In these Application Notes, the Automatic Route Selection (ARS) feature will be used to route outbound calls via the SIP trunk to PAETEC. In the sample configuration, the single digit 9 is used as the ARS access code. Avaya telephone users will dial 9 to reach an "outside line". The common configuration is illustrated below with little elaboration. **Figure 29** shows the **change dialplan analysis** command. Observe that a dialed string beginning with 9 of length 1 is a feature access code (**fac**). The use of 5 digit extensions with first digit 2 can also be observed.

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

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     0            1    dac
     1            4    dac
     2            5    ext
     8            1    fac
     9            1    fac
     *            3    fac
     #            3    fac
```

Figure 29: Dialplan Analysis Form

Use the **change feature-access-codes** command to configure or observe 9 as the ARS access code, as shown in **Figure 30**.

```
change feature-access-codes                                    Page   1 of   7
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: *70
         Abbreviated Dialing List2 Access Code: *80
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA: *90    All: *72    Deactivation: #73
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code: *77
```

Figure 30: Feature Access Codes Form

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. A small sampling of dial patterns is illustrated here. Further administration of ARS is beyond the scope of these Application Notes. Consult references [2] and [3]. During compliance testing, domestic long-distance calls, international calls, 411 calls, 911 calls, dial 0, and 0+ calls were all routed successfully through the PAETEC network via ARS.

Figure 31 shows example **ars analysis** configuration for numbers such as 1-732-852-16XX. Calls are sent to Route Pattern 2, which will contain the Outbound PSTN SIP Trunk Group to PAETEC.

```
change ars analysis 173                                         Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    1

          Dialed            Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    17326870755           11   11    2        fnpa         n
    17328521243           11   11    2        fnpa         n
    173285216             11   11    2        natl         n
    17328522496           11   11    2        fnpa         n
    17328523500           11   11    2        fnpa         n
```

Figure 31: ARS Analysis Form 1732 Numbers

Figure 32 shows example **ars analysis** configuration for a service number such as 411. Again, calls are sent to Route Pattern 2.

```
change ars analysis 411                                         Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    1

          Dialed            Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    411                   3    3     2        svcl         n
```

Figure 32: ARS Analysis Form 411

Figure 33 shows example **ars analysis** configuration for certain operator calls. Again, calls are sent to Route Pattern 2.

```
change ars analysis 0                                           Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    1

          Dialed            Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    0                     1    1     2        op           n
    0                     11   11    2        op           n
```

Figure 33: ARS Analysis Form 0

Use the **change route-pattern** command to add the SIP trunk group to the route pattern that ARS selects, as shown in **Figure 34**. In this configuration, route pattern 2 is used to route calls to trunk group 2. As can be observed, Look-Ahead Routing (LAR) can optionally be used to allow calls to complete automatically using a different trunk group, should the SIP Trunk Group to PAETEC (or Avaya SES) be non-responsive, or if specific SIP messages are received from PAETEC (or Avaya SES) in response to an outbound PSTN call attempt. See reference [13] for more information on LAR. In the sample configuration, trunk group 22 is an ISDN-PRI trunk group to another system, used as a second choice in the route pattern.

```
change route-pattern 2                                             Page   1 of   3
                    Pattern Number: 2   Pattern Name: To PAETEC
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                                DCS/ IXC
    No          Mrk Lmt List Del  Digits                                  QSIG
                             Dgts                                         Intw
 1: 2    0       1                                                         n   user
 2: 22   0       1                *9                                       n   user
 3:                                                                        n   user
 4:                                                                        n   user
 5:                                                                        n   user
 6:                                                                        n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                            next
 2: y y y y y n  n            rest                                            none
 3: y y y y y n  n            rest                                            none
 4: y y y y y n  n            rest                                            none
 5: y y y y y n  n            rest                                            none
 6: y y y y y n  n            rest                                            none
```

Figure 34: Route-Pattern Containing Outbound PSTN SIP Trunk Group

Step 9: Configure Incoming Digit Translation

This step configures the settings necessary to map incoming DID calls to the proper Avaya Communication Manager extension(s). The incoming digits sent in the INVITE message from PAETEC can be manipulated as necessary to route calls to the desired extension. In the examples used in the compliance testing, the incoming DID numbers provided by PAETEC do not have any correlation to the internal extensions assigned within Avaya Communication Manager. Thus, all incoming digits are deleted and replaced by the assigned extension number.

To create a fully mapped extension number as shown in **Figure 35**:

- Open the **Incoming Call Handling Treatment** form for the Incoming PSTN SIP trunk group configured in **Figure 21**, in this case, trunk group 3.
- For each extension assigned a DID number from PAETEC, enter **10** into **Called Len**, **all** into **Del**, and the entire **10 digit DID number** into the **Called Number** field.
  Enter an Avaya Communication Manager extension number into the **Insert** field.

```
change inc-call-handling-trmt trunk-group 3                        Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called      Del Insert
 Feature        Len      Number
 public-ntwrk    10 2132260033       all 20004
 public-ntwrk    10 2132260034       all 20006
```

Figure 35: Incoming Call Handling Treatment – Full Extension Mapping

If the customer's extension numbering plan aligns with the DID numbers in some meaningful way (e.g., if the final DID digits match the extension), it is not necessary to define an entry for each DID number. As a hypothetical example, assume a PBX dial plan that used the 5 digit extensions 60000 thru 69999, and assume PAETEC provided DID numbers of 732-626-0000 thru 9999. The incoming number translation could be done similar to **Figure 36**. Note that the Called Number entry in this case represents the common matching portion applicable to all incoming numbers. Thus, one entry matches all numbers in the assigned DID block from PAETEC.

```
change inc-call-handling-trmt trunk-group 3                       Page   1 of  30
                              INCOMING CALL HANDLING TREATMENT
        Service/       Called    Called        Del  Insert
        Feature         Len      Number
        public-ntwrk     10      732626          5
```

Figure 36: Incoming Call Handling Treatment – Hypothetical Extension Mapping

Step 10: 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 such as Avaya 9600-Series SIP Telephones, and assumes the preceding SIP Trunk configuration to have been completed. SIP telephones are optional and not required to use the PAETEC Dynamic IP SIP Trunk Service.

Step 1: **Assign a Station**
Assign a station as shown in **Figure 37**. This example uses an Avaya one-X 9620 Deskphone. Using the **add station** command from the SAT:

- Set the station **Type** to the value "9620".
- Enter a **Name** for the user of the station.
- The **Security Code** may be left blank for SIP OPS extensions, since SIP Telephones will register with Avaya SES.

The remaining fields are configured per normal station administration. Note that the Class of Restriction (**COR**) and Class of Service (**COS**) defined in Avaya Communication Manager will govern features and call restrictions that apply to this station.

```
add station 20000                                              Page   1 of   6
                                     STATION

Extension: 20000                         Lock Messages? n              BCC: 0
     Type: 9620                          Security Code:                TN: 1
     Port: S00000                      Coverage Path 1:                COR: 1
     Name: John Doe                    Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 20000
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english
```

Figure 37: Station Administration – Page 1

On Page 4 of the **Station** form, configure at least 3 call appearances under the Button Assignments section for the SIP telephone, and any other supported telephone button features, as shown in **Figure 38**.

```
add station 20000                                            Page   4 of   6
                                     STATION
 BUTTON ASSIGNMENTS
 1: call-appr                            4: ec500      Timer? n
 2: call-appr                            5: extnd-call
 3: call-appr                            6: no-hld-cnf
```

Figure 38: Station Administration – Page 4

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

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 in turn direct the call to the registered SIP telephone. On the **Off-PBX-Telephone Station-Mapping** form shown in **Figure 39**:

- 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 SES match the extensions of the corresponding stations on Avaya Communication Manager.
- Set the **Trunk Selection** field to *1*, which is the number assigned to the SIP OPS trunk group. This trunk group number was previously defined in **Figure 16**.
- 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 20000                  Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 20000             OPS           -     20000              1          1
```

Figure 39: Stations with Off-PBX Telephone Integration – Page 1

On Page 2, set the **Call Limit** field to the number of calls that may be active 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. The default values for other fields may be accepted.

```
change off-pbx-telephone station-mapping 20000                  Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged        Location
 Extension     Limit      Mode         Allowed     Calls
 20000          3       both           all         none
```

Figure 40: 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.

3.3 Configuration of Non-G.729A SIP Endpoints

The PAETEC Dynamic IP SIP Trunk Service supports G.729A, but not G.729B. However, the Avaya 4600-Series SIP telephones support G.729B, but do not support G.729A. As a result, "shuffling" to ip-direct media must not occur for calls involving Avaya 4600-Series SIP Telephones and the PAETEC Dynamic IP SIP Trunk Service. In the compliance testing, calls involving Avaya 4600-Series SIP Telephones successfully communicated using G.711MU to the Avaya TN2601AP IP Media Resource card, which in turn presented G.729A on the leg of the connection facing the PAETEC network.

4. Configure Avaya SIP Enablement Services

This section covers the administration of Avaya SIP Enablement Services (SES). Avaya SES 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. For additional information on installation tasks, refer to [5].

This section is divided into two parts: **Section 4.1** provides the steps necessary to configure a SIP trunk to PAETEC. **Section 4.2** provides the steps necessary to complete the administration for optional SIP endpoints.

4.1 SIP Trunking to PAETEC

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 the 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 41**.

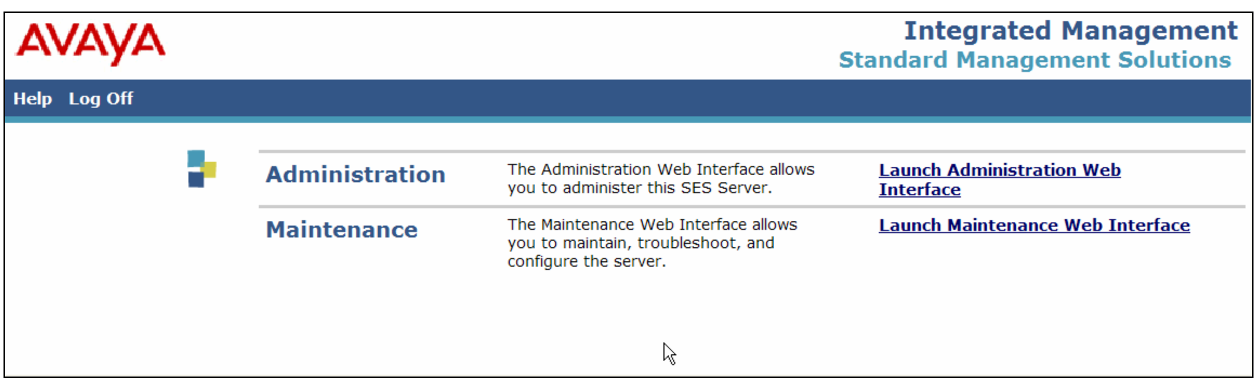

Figure 41 - Avaya SES Main Screen

The SES administration home screen shown in **Figure 42** will be displayed.

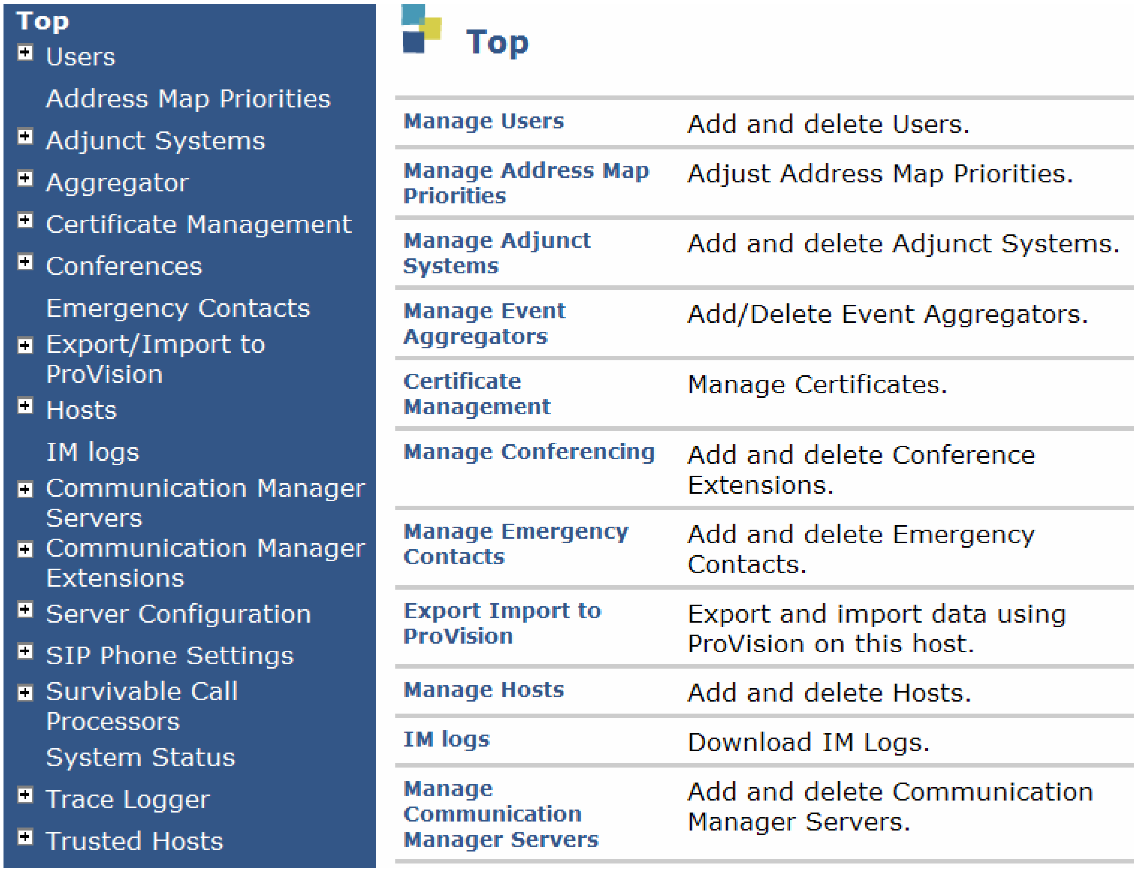

Figure 42: Avaya SES Administration Home Page

Step 2: Verify 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 network properties configured during the installation process. In the **System Properties** screen, verify the **SIP Domain** name assigned to Avaya SIP Enablement Services. This domain should match the domain configured in Avaya Communication Manager for the network region for local users (**Figure 7**) and the SIP signaling group to Avaya SES for SIP OPS Telephones (**Figure 16**).

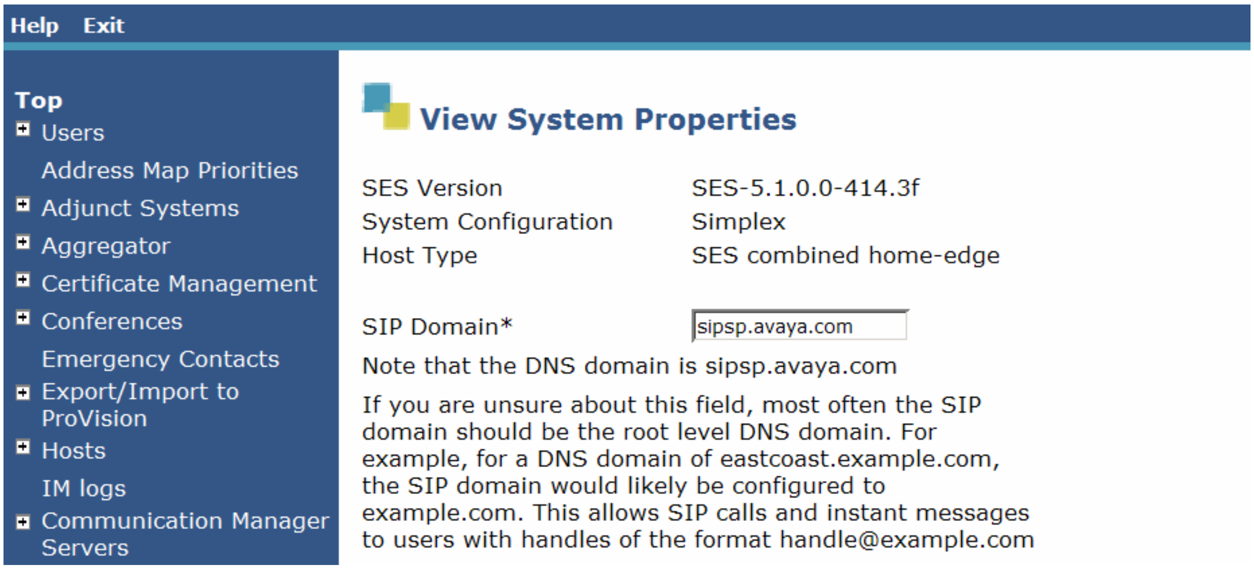

Figure 43: System Properties Showing SIP Domain

Step 3: Verify the Avaya SES Host Information

Verify the Avaya SES Host information using the **Edit Host** page. In these Application Notes, the Avaya SES **Host Type** is a combined ***home/edge***. This means that both the PAETEC Dynamic IP SIP Trunk Service and Avaya Communication Manager are contacting the same SES. Display the **Edit Host** page (**Figure 44)** by following the **Hosts** link in the left navigation pane and then clicking on the **Edit** option under the **Commands** section of the **List Hosts** screen.

On the **Edit Host** screen shown in **Figure 44**:

- Verify the **Host IP Address** of this combined SES Home/Edge server.
- Verify that the **UDP**, **TCP** and **TLS** checkboxes are enabled as **Listen Protocols**.
- Verify that **TLS** is selected via **Link Protocols**.
- Default values for the remaining fields may be used.

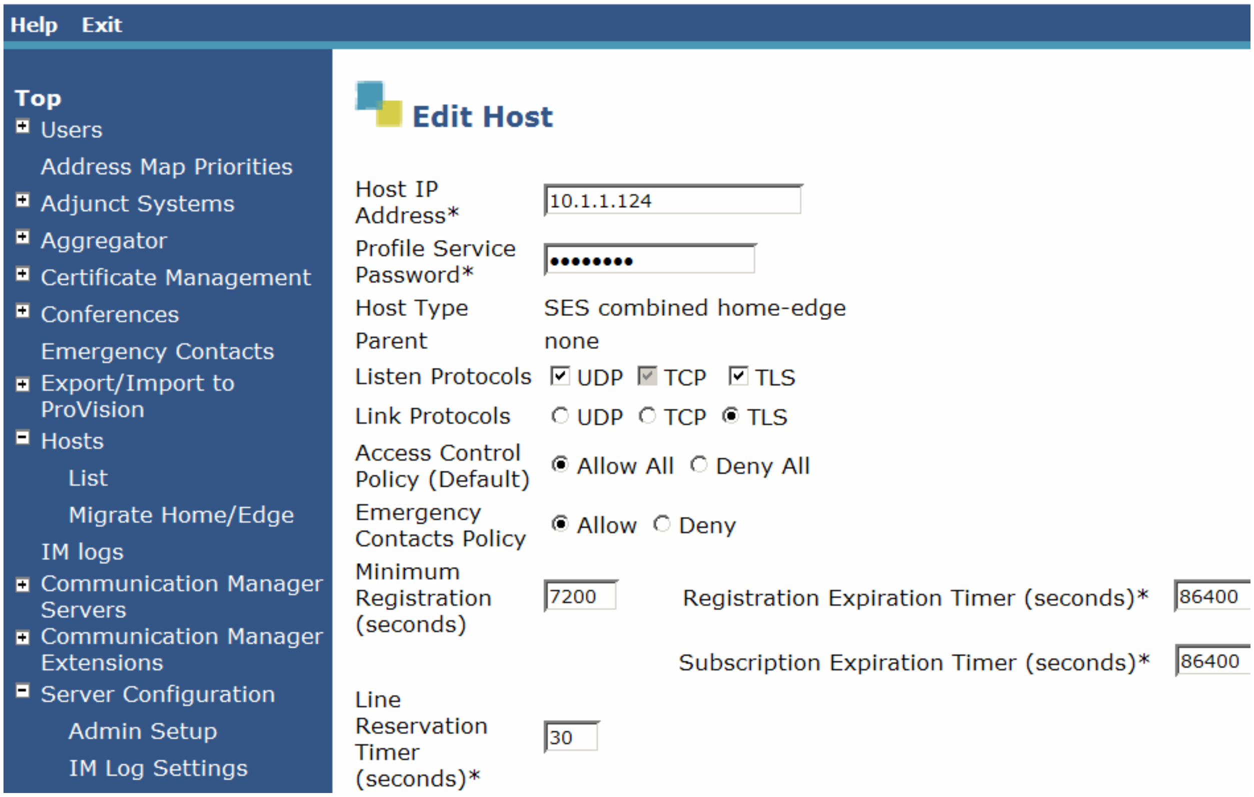

Figure 44: Edit Host

Step 4: **Add Avaya Communication Manager Server**
Expand the **Communication Manager Servers** option in the Administration web interface, and select **Add**. This step will create the Avaya SES side of the SIP trunk previously created in Avaya Communication Manager.

In the **Add Communication Manager Server Interface** screen, enter a descriptive name in the **Communication Manager Server Interface Name** field (e.g., "S8720-CLAN"). The IP Address of the single Home/Edge SES Server is automatically entered in the **Host** field. Select TLS (Transport Link Security) for the **SIP Trunk Link Type**. Enter the IP address of the C-LAN board used in the definition of the SIP signaling group to SES (**Figure 16**) in the **SIP Trunk IP Address** field. In alternate configurations such as those using the Avaya S8300 Server, this may be the IP address of the Avaya S8300 Server. Scroll to the bottom, and click **Add** (not shown).

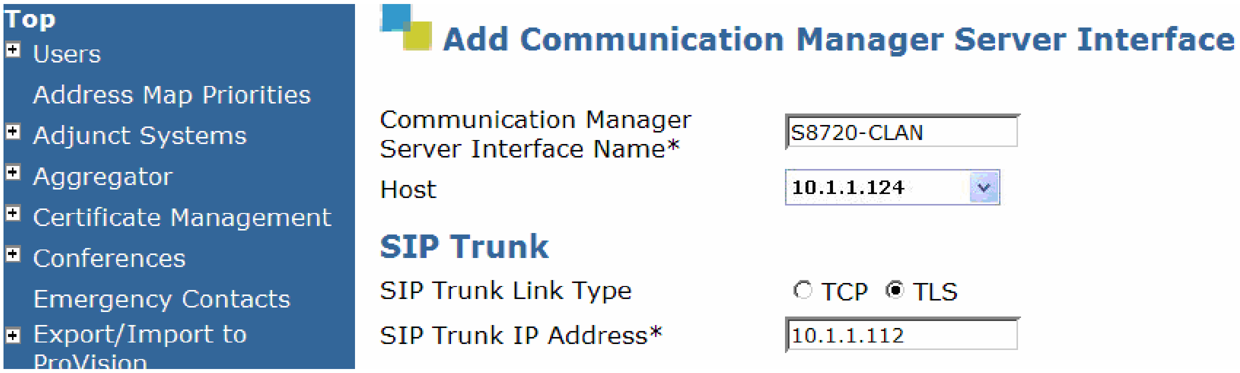

Figure 45: Add Communication Manager Server Interface

Step 5: Specify Address Maps

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 Communication Manager 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 Communication Manager 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 same Avaya SES.

In these Application Notes, only incoming calls from the PSTN require a Communication Manager address map entry. Calls originated by Avaya SIP telephones are automatically routed to the proper Avaya Communication Manager by the assignment of an Avaya Communication Manager Server extension to that phone user.

For the PAETEC Dynamic IP SIP Trunk 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 would be:

sip:2132260034@10.1.1.124;user=phone;

The user portion in this case is the 10 digit DID number "2132260034". One or more address maps can be created to match the DID numbers assigned to the customer by PAETEC. The SES will forward the messages based on the matching patterns to the appropriate C-LAN interface controlled by the S8720 Server.

To configure a **Communication Manager Server Address Map**:

- Select **Communication Manager Servers** in the left pane of the Administration web interface.
- Click on the **Map** link associated with the appropriate server.

- Click on the **Add Map In New Group** link.

In the screen shown in **Figure 46:**

- 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, example DID numbers provided by PAETEC are 213-226-0033 and 213-226-0034. An example pattern specification (without the double quotes) for these DID numbers is: “^sip:213226003[34]”. URIs beginning with “sip:213226003” followed by either the digit 3 or 4 will match the pattern and be routed to the interface defined for the C-LAN associated with this Communication Manager Server. Appendix B provides an overview of the syntax for address map patterns.
- Click the **Add** button once the form is completed.

Figure 46: Communication Manager Server Address Map

After adding the address map, the **List Communication Manager Server Address Map** screen will appear, as shown in **Figure 47**.

Figure 47: List Communication Manager Server Address Map

When the **Communication Manager Server Address Map** is added, a **Contact** is created automatically. For the **Communication Manager Server Address Map** added in **Figure 45**, the following contact was created:

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

The contact specifies the IP address of the C-LAN 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: Configure the PAETEC SIP Network Element(s) as Trusted Host(s)
The IP addresses provided by PAETEC for SIP network elements must be added as trusted hosts to the Avaya SES. For a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the designated IP address. If multiple SIP proxies are used in the PAETEC network to route calls to the Avaya SES in the enterprise, the IP address of each must be added as a trusted host.

Expand **Trusted Hosts** from the lower left of the SES Administration page (shown in **Figure 42**). Click **Add**. In the **Add Trusted Host** screen shown in **Figure 48**, enter the IP Address provided by PAETEC for the PAETEC network element in the **IP Address** field. (Recall that the actual IP Addresses used during compliance-testing are not included in these Application Notes). In the **Host** drop-down, select the Host corresponding to the Avaya SES for which the trust relationship must exist. Click **Add**.

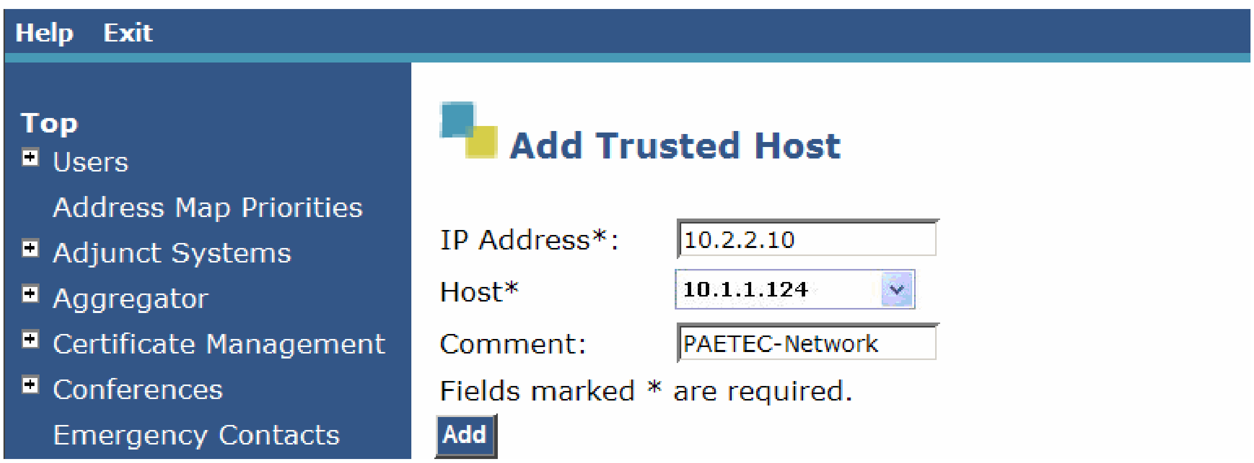

Figure 48: Adding a Trusted Host

A screen like **Figure 49** will appear. Click **Continue**.

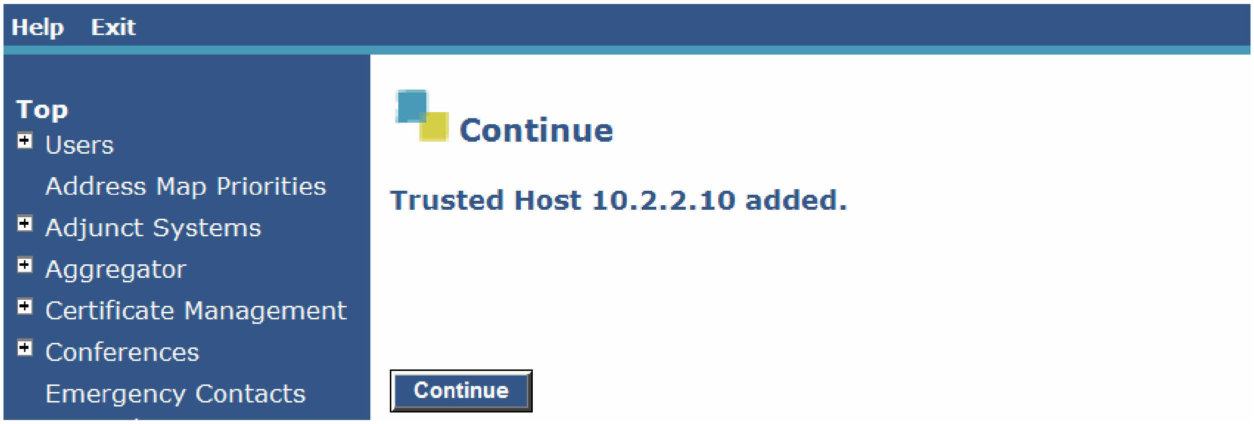

Figure 49: Continue Adding a Trusted Host

4.3 Configuration for Optional SIP Telephones

This section provides basic instructions for completing the SES administration necessary to support optional Avaya SIP telephones.

Step 1: Add a SIP User

In Avaya SES Administration, expand **Users**. Click **Add**. In the **Add User** screen shown in **Figure 50**:

- Enter the extension of the SIP user 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 SES hosting the domain for this user.
- Enter the **First Name** and **Last Name** of the user.

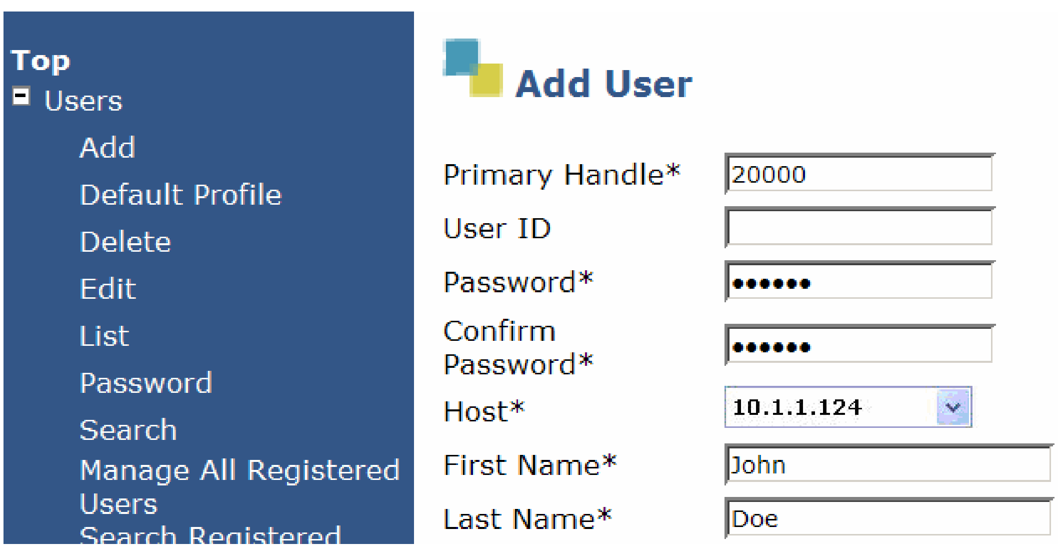

Figure 50: Add User – User Information

Scroll to the bottom of the **Add User** page, and select the **Add Communication Manager Extension** checkbox as shown in **Figure 51**. Click **Add**.

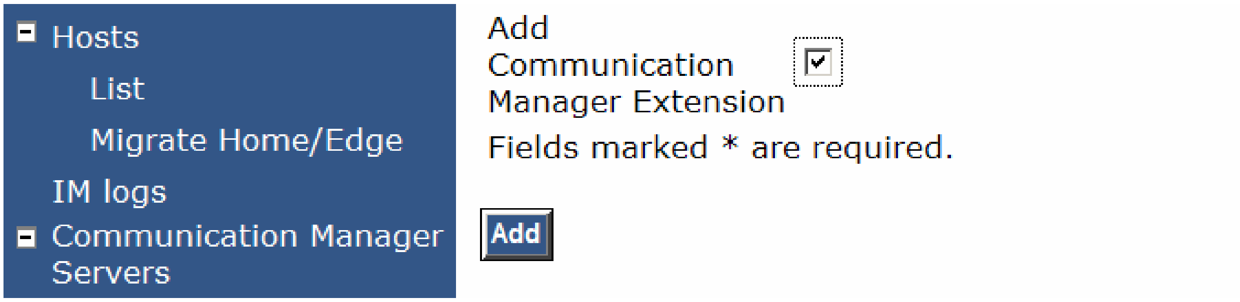

Figure 51: Add User – Add Communication Manager Extension Area

Press **Continue** at the confirmation screen.

Step 2: Specify Corresponding Avaya Communication Manager Extension
The SIP phone handle must now be associated with the corresponding extension in Avaya Communication Manager. In the **Add Communication Manager Server Extension** screen shown in **Figure 52**:

- Enter the **Extension** configured on Avaya Communication Manager, configured in **Figure 37**.
- From the drop-down, select the **Communication Manager Server** associated with this extension.
- Click **Add**.

Figure 52: Add Communication Manager Extension

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

5. PAETEC Services Configuration

To use PAETEC Communications Dynamic IP SIP Trunk Service, a customer must request service from PAETEC using their sales processes. The process can be started by contacting PAETEC Communications via the 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. PAETEC Communications provided the following information for the compliance testing: IP address of the PAETEC Communications SIP proxy/SBC, Direct Inward Dialed (DID) numbers. This information was used to complete the Avaya Communication Manager and Avaya SIP Enablement Services configuration discussed in the previous sections.

6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunk interoperability between the PAETEC Dynamic IP SIP Trunk Service and an Avaya IP Telephony Solution. This section covers the general test approach and the test results.

6.1. General Test Approach

A simulated enterprise site using an Avaya IP telephony solution was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the commercially available Dynamic IP SIP Trunk Service provided by PAETEC.

The compliance test included the following:

- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by PAETEC. Incoming PSTN calls were made to H.323, digital, analog, and SIP telephones at the enterprise.
- Outgoing calls from the enterprise site were completed via PAETEC to PSTN destinations. Outgoing calls from the enterprise to the PSTN were made from H.323, digital, analog, and SIP telephones.
- Various outbound call types including: long distance, international, directory assistance (411), operator (0, 0+), and 911.
- Calls using G.729A, G.711MU, and G.711A coders.
- Fax calls completed using the G.711MU coder. The PAETEC Dynamic IP SIP Trunk Service does not support T.38.
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, conference, analog call waiting, etc.
- Off-net call forwarding and extension to cellular, when the call arrived from the SIP Trunk from PAETEC, or when the call forwarding destination and extension to cellular mobile number routed out the SIP Trunk to PAETEC, or both.
- Caller ID Presentation and Caller ID Restriction (See Section 6.2.1).

- Avaya IP Softphone in both “Road Warrior” and “Telecommuter” modes, where incoming PSTN calls arrived from PAETEC, or the telecommute number routed out the SIP Trunk to PAETEC, or both.
- Direct IP-to-IP media (also known as “shuffling”) with SIP and H.323 telephones.
- PAETEC network-based maintenance via periodic transmission of SIP OPTIONS messages by PAETEC requiring Avaya response. PAETEC offers trunk group overflow for PSTN calls destined for an enterprise location that has not responded to repeated SIP OPTIONS heartbeats.
- Avaya Communication Manager Look-Ahead Routing for SIP Trunks, enabling enterprise based trunk group overflow for calls destined for the PSTN, should the PAETEC Dynamic IP Trunk SIP Service be unresponsive, or respond with specific SIP error messages, as detailed in reference [13].

6.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the PAETEC Dynamic IP SIP Trunk Service. However, the following issue was observed.

6.2.1 Calling Number Restriction

Issue Observed

When the Avaya “cpn-blk” feature button was used, a call routed to the PAETEC SIP Trunk was not marked for privacy in the SIP INVITE message sent by Avaya Communication Manager. The caller id was displayed to the called PSTN user. A product modification request has been entered (defsw082674).

Discussion/Workaround

Other means exist to achieve privacy for caller id for specific users, but these alternate methods are associated with withholding or restricting caller id for a given user or trunk for *all* calls. For example, at the user level, individual privacy can be achieved using the ”public unknown numbering” form. However, such privacy would apply to the user for all calls, or all calls using a specific trunk. The “cpn-blk” feature button and corresponding access code are intended to enable a user whose calls normally allow caller id presentation to restrict presentation of caller id for a specific call. This Avaya Communication Manager capability was not functioning properly for SIP trunks in the software versions used for testing.

From a testing work-around perspective (not an appropriate end-user workaround), another approach to including the “Privacy: Id” designation in an outbound Avaya SIP INVITE was tested. If the telephone user is placed in a Class of Restriction marked to mask the Calling Party Number, privacy for calls using the SIP trunk is also achieved. In this case, Avaya Communication Manager includes the “Privacy: Id” designation as well as other “anonymous” indications in the SIP INVITE message, and the caller id is not displayed to the called PSTN user when the call is routed through the PAETEC Dynamic IP SIP Trunk Service.

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 calls using the PAETEC Dynamic IP SIP Trunk Service.

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 end an active call by hanging up.
4. Verify that an endpoint at the enterprise site can end an active call by hanging up.

8. Support

For technical support on PAETEC Communications Dynamic IP SIP Trunk services, contact PAETEC Communications Customer Service by calling 877-340-2600 or by sending email to customerservice@PAETEC.com. Include the customer account number in the communication.

9. Conclusion

These Application Notes describe the configuration steps enabling customers using Avaya Communication Manager and Avaya SIP Enablement Services to connect to the PSTN via the PAETEC Dynamic IP SIP Trunk Service. The PAETEC Dynamic IP SIP Trunk Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. The PAETEC Dynamic IP SIP Trunk Service 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. Additional Avaya product documentation is available at http://support.avaya.com.

[1] Application Notes for Configuring SIP Trunking between the PAETEC Communications IPATH Service and an Avaya SIP Telephony Solution, Issue 1.0, 11/20/2006

http://www.avaya.com/master-usa/en-us/resource/assets/applicationnotes/paetec_appnote.pdf

[2] *Administrator Guide for Avaya Communication Manager*, January 2008, Document Number 03-300509.

[3] *Feature Description and Implementation for Avaya Communication Manager*, January 2008, Document Number 555-245-205

[4] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 4.0*, Feb 2007, Issue 10, Document Number 210-100-700.

[5] *SIP Enablement Services Implementation Guide, Jan* 2008, Document Number 16-300140

[6] *SIP Support in Avaya Communication Manager Running on Avaya Servers, Jan* 2008, Document Number 555-245-206.

[7] *4600 Series IP Telephone LAN Administrator Guide*, October 2007, Document Number 555-233-507

[8] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide Release 2.0*, May 2008, Document Number 16-300698

[9] *Avaya one-X Deskphone SIP for 9600 Series IP Telephones Administrator Guide Release 2.0*, Dec 2007, 16-601944

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

[11] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, http://www.ietf.org/

[12] RFC 4244, *An Extension to the Session Initiation Protocol (SIP) for Request History Information*, http://www.ietf.org/

[13] Sample Configuration for SIP Private Networking and SIP Look-Ahead Routing Using Avaya Communication Manager Application Notes

APPENDIX A: Sample SIP INVITE Messages

This appendix displays example SIP INVITE messages for inbound and outbound calls. Customers may use these INVITE messages for comparison and troubleshooting purposes. Differences in these messages may indicate different configuration options selected.

The example message below was sent by PAETEC to the Avaya SES at the enterprise site. The call is from a cellular telephone user to the PAETEC-provided DID 213-226-0034.

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

```
Session Initiation Protocol
    Request-Line: INVITE sip:2132260034;npdi@10.1.1.124:5060;user=phone SIP/2.0
        Method: INVITE
        Resent Packet: False
    Message Header
        Via: SIP/2.0/UDP 10.2.2.10:5060;branch=z9hG4bK000snd202o6hff862040.1
        From: "Unavailable"<sip:7326870755@10.2.2.10:5060;isup-oli=61;user=phone;interop-
sip-tg2plex=interop-sip-tg2plex-tj3qs4euhjvn6>;tag=127.0.0.15060+1+56840003+4a89678c
            SIP Display info: "Unavailable"
            SIP from address: sip:7326870755@10.2.2.10:5060
            SIP tag: 127.0.0.15060+1+56840003+4a89678c
        To: <sip:2132260034@10.1.1.124:5060;user=phone>
            SIP to address: sip:2132260034@10.1.1.124:5060
        CSeq: 1012264661 INVITE
        Expires: 180
        Min-SE: 1800
        Session-Expires: 1800
        Supported: timer
        Request-Disposition: fork, parallel
        Allow: INVITE, BYE, REGISTER, ACK, OPTIONS, CANCEL, SUBSCRIBE, NOTIFY,
PRACK, INFO, REFER, UPDATE
        Call-ID: A0A52390-5057CC@10.254.1.7
        P-Asserted-Identity: "Unavailable"<sip:7326870755@10.254.1.7;user=phone>
        Privacy: none
        Max-Forwards: 50
        Contact: "Unavailable" <sip:7326870755@10.2.2.10:5060;interop-sip-tg2plex=interop-sip-
tg2plex-tj3qs4euhjvn6;transport=udp>
            Contact Binding: "Unavailable" <sip:7326870755@10.2.2.10:5060;interop-sip-
tg2plex=interop-sip-tg2plex-tj3qs4euhjvn6;transport=udp>
                URI: "Unavailable" <sip:7326870755@10.2.2.10:5060;interop-sip-tg2plex=interop-
sip-tg2plex-tj3qs4euhjvn6;transport=udp>
                    SIP Display info: "Unavailable"
                    SIP contact address: sip:7326870755@10.2.2.10:5060
        Content-Type: application/sdp
        Content-Length: 333
```

Message body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 3427817421 3427817421 IN IP4 10.2.2.10
Owner Username: -
Session ID: 3427817421
Session Version: 3427817421
Owner Network Type: IN
Owner Address Type: IP4
Owner Address: 10.2.2.10
Session Name (s): -
Connection Information (c): IN IP4 10.2.2.10
Connection Network Type: IN
Connection Address Type: IP4
Connection Address: 10.2.2.10
Time Description, active time (t): 0 0
Session Start Time: 0
Session Stop Time: 0
Media Description, name and address (m): audio 20018 RTP/AVP 0 8 18 98 101
Media Type: audio
Media Port: 20018
Media Proto: RTP/AVP
Media Format: ITU-T G.711 PCMU
Media Format: ITU-T G.711 PCMA
Media Format: ITU-T G.729
Media Format: 98
Media Format: 101
Media Attribute (a): ptime:20
Media Attribute Fieldname: ptime
Media Attribute Value: 20
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute Fieldname: rtpmap
Media Attribute Value: 0 PCMU/8000
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute Fieldname: rtpmap
Media Attribute Value: 8 PCMA/8000
Media Attribute (a): rtpmap:18 G729/8000
Media Attribute Fieldname: rtpmap
Media Attribute Value: 18 G729/8000
Media Attribute (a): rtpmap:98 clearmode/8000
Media Attribute Fieldname: rtpmap
Media Attribute Value: 98 clearmode/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute Fieldname: rtpmap
Media Attribute Value: 101 telephone-event/8000
Media Attribute (a): fmtp:101 0-15

Media Attribute Fieldname: fmtp
Media Attribute Value: 101 0-15
Media Attribute (a): fmtp:18 annexb=no
Media Attribute Fieldname: fmtp
Media Attribute Value: 18 annexb=no
Media Attribute (a): silenceSupp:off - - - -
Media Attribute Fieldname: silenceSupp
Media Attribute Value: off - - - -

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

This trace corresponds to the initial INVITE for an outbound call from an H.323 IP Telephone with extension 20004 and name “H.323 9640” to PSTN destination 1-732-852-1639. The codec requested for the call is G.729A. At the time of this trace, the configuration that explicitly configures the telephone event payload type to 101 (**Figure 20**) was not yet performed. This is included so that the default behavior for telephone events (127) can be observed. Recall that the actual IP Addresses have been changed. All IP Addresses in the trace below are shown in the sample configuration screens in these Application Notes except 10.1.1.60, which is the IP Address of a TN2601AP IP Media Resource card.

Session Initiation Protocol
Request-Line: INVITE sip:17328521639@10.2.2.10 SIP/2.0
Method: INVITE
Resent Packet: False
Message Header
Accept-Language: en
Call-ID: 80e0ff151075dd132154891b2ff00
CSeq: 1 INVITE
From: "H.323 9640"
<sip:7328520004@sipsp.avaya.com:5061>;tag=80e0ff151075dd131154891b2ff00
SIP Display info: "H.323 9640"
SIP from address: sip:7328520004@sipsp.avaya.com:5061
SIP tag: 80e0ff151075dd131154891b2ff00
Record-Route: <sip:10.1.1.124:5060;lr>,<sip: 10.1.1.112:5061;lr;transport=tls>
To: "17328521639" <sip:17328521639@10.2.2.10>
SIP Display info: "17328521639"
SIP to address: sip:17328521639@10.2.2.10
Via: SIP/2.0/UDP
10.1.1.124:5060;branch=z9hG4bK83838303030356562ecd.0,SIP/2.0/TLS
10.1.1.112;psrrposn=2;received=10.1.1.112;branch=z9hG4bK80e0ff151075dd133154891b2ff00
Content-Length: 187
Content-Type: application/sdp
Contact: "H.323 9640" <sip:7328520004@10.1.1.112:5061;transport=tls>
Contact Binding: "H.323 9640" <sip:7328520004@10.1.1.112:5061;transport=tls>
URI: "H.323 9640" <sip:7328520004@10.1.1.112:5061;transport=tls>
SIP Display info: "H.323 9640"

```
            SIP contact address: sip:7328520004@10.1.1.112:5061
      Max-Forwards: 68
      User-Agent: Avaya CM/R015x.01.0.414.3
      Allow: 
INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS,INFO,PUBLI
SH
      Supported: 100rel,timer,replaces,join,histinfo
      Alert-Info: <cid:internal@10.2.2.10>;avaya-cm-alert-type=internal
      Min-SE: 1800
      Session-Expires: 1800;refresher=uac
      P-Asserted-Identity: "H.323 9640" <sip:7328520004@sipsp.avaya.com:5061>
      History-Info: <sip:17328521639@10.2.2.10>;index=1,"17328521639" 
<sip:17328521639@10.2.2.10>;index=1.1
   Message body
      Session Description Protocol
         Session Description Protocol Version (v): 0
         Owner/Creator, Session Id (o): - 1 1 IN IP4 10.1.1.112
            Owner Username: -
            Session ID: 1
            Session Version: 1
            Owner Network Type: IN
            Owner Address Type: IP4
            Owner Address: 10.1.1.112
         Session Name (s): -
         Connection Information (c): IN IP4 10.1.1.60
            Connection Network Type: IN
            Connection Address Type: IP4
            Connection Address: 10.1.1.60
         Bandwidth Information (b): AS:64
            Bandwidth Modifier: AS
            Bandwidth Value: 64
         Time Description, active time (t): 0 0
            Session Start Time: 0
            Session Stop Time: 0
         Media Description, name and address (m): audio 26724 RTP/AVP 18 127
            Media Type: audio
            Media Port: 26724
            Media Proto: RTP/AVP
            Media Format: ITU-T G.729
            Media Format: 127
         Media Attribute (a): rtpmap:18 G729/8000
            Media Attribute Fieldname: rtpmap
            Media Attribute Value: 18 G729/8000
         Media Attribute (a): fmtp:18 annexb=no
            Media Attribute Fieldname: fmtp
            Media Attribute Value: 18 annexb=no
```

Media Attribute (a): rtpmap:127 telephone-event/8000
    Media Attribute Fieldname: rtpmap
    Media Attribute Value: 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, matched 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 1+ 10 digit number 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.

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