Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking Using Packet One Networks SIP Trunk Service and Avaya IP Telephony Solution – 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the Packet One Networks SIP Trunk Service and an Avaya IP telephony solution. The Avaya solution consists of Avaya Aura™ SIP Enablement Services, Avaya Aura™ Communication Manager and various Avaya H.323, digital and analog endpoints.

Packet One Networks 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 Packet One Networks SIP Trunk service and an Avaya IP telephony solution. The Avaya solution consists of Avaya Aura™ SIP Enablement Services, Avaya Aura™ Communication Manager, and various Avaya H.323, digital and analog endpoints. Customers using this Avaya IP telephony solution with the Packet One Networks 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. **Figure 1** illustrates an example Avaya IP telephony solution connected to the Packet One Networks 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 S8300c Server running Avaya Aura™ Communication Manager Release 5.2
- Avaya G450 Media Gateway and associated hardware
- Avaya S8500 Server running Avaya Aura™ SIP Enablement Services (SES) Release 5.2
- Avaya 9600-Series IP telephones (configured for SIP and H.323 protocols)
- Avaya digital telephones
- Analog telephones and fax machines

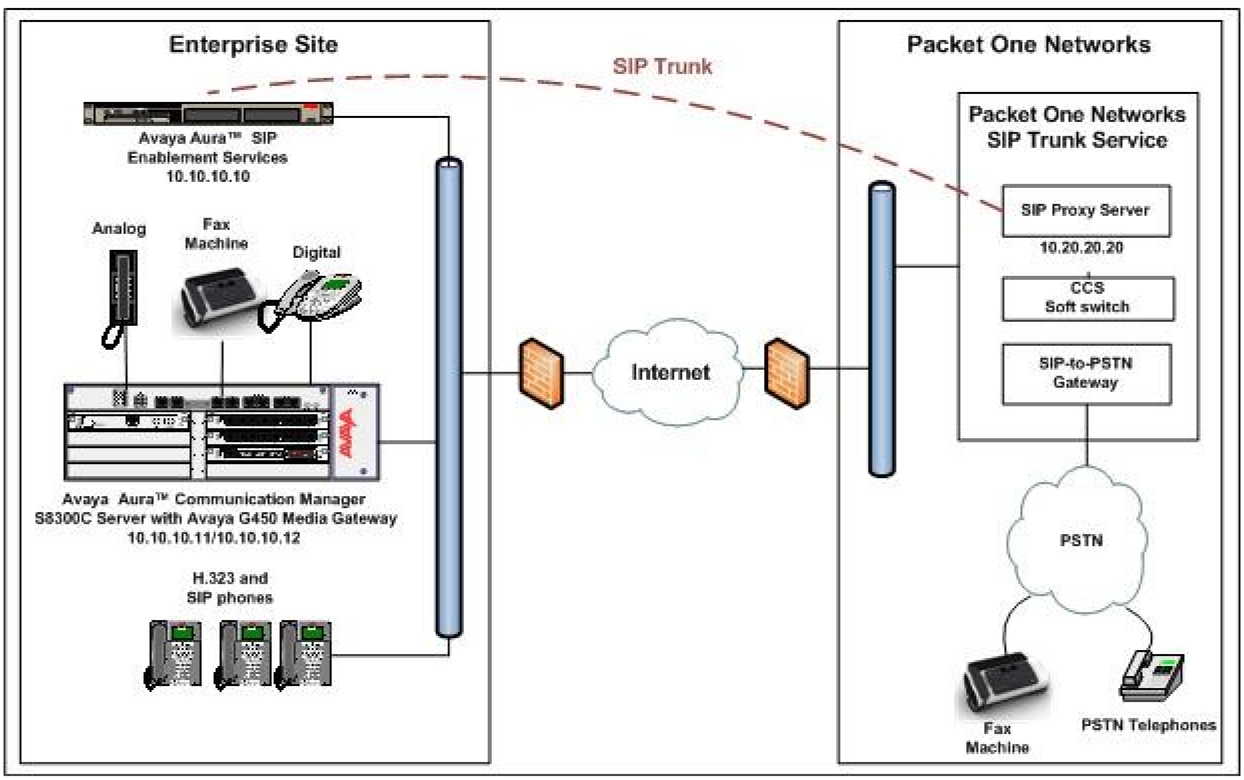

Figure 1: Avaya IP Telephony Network using Packet One Networks SIP Trunk Service

1.1. Interoperability Compliance Testing

The interoperability compliance test focused on the ability for the Packet One solution to interoperate with the Avaya VOIP solution. The following is a summary of the feature and serviceability testing that was undertaken.

- Basic Calls, which including calling/connected party name/number display and restriction
- Codec Negotiation
- Call Hold and resume
- Conference
- Call forwarding
- Call Transfer
- Voice mail box access and message retrieval
- DTMF transmission
- Fax calls using T.38

1.2. Support

For technical support on Packet One SIP Trunk services, contact Packet One Customer Care by calling 1-300-800-888 or by sending email to careline@packet-1.com.

1.3. 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 or H.323 telephone at the enterprise site, as described below.

1. A user on the PSTN dials a DID number provided by Packet One Networks which is assigned to Avaya Aura™ Communication Manager telephone at the enterprise site. The PSTN routes the call to Packet One Networks. Packet One Networks then routes the DID number to the assigned customer.
2. Based on the DID number, Packet One Networks offers the call to SIP Enablement Services using SIP signaling messages sent over the converged access facility. The assignment of the DID number and the address of the SIP Enablement Services are established during the ordering and provisioning of the service.
3. SIP Enablement Services routes the call to Avaya Aura™ Communication Manager, also using a SIP trunk.
4. Avaya Aura™ Communication Manager rings the analog, digital, or H.323 telephone, as shown in **Step 4** below.

Or

5. If the inbound call is to a SIP extension at the enterprise, Avaya Aura™ Communication Manager transmits the appropriate SIP signaling via SIP Enablement Services to the SIP telephone, as shown by arrow 5.

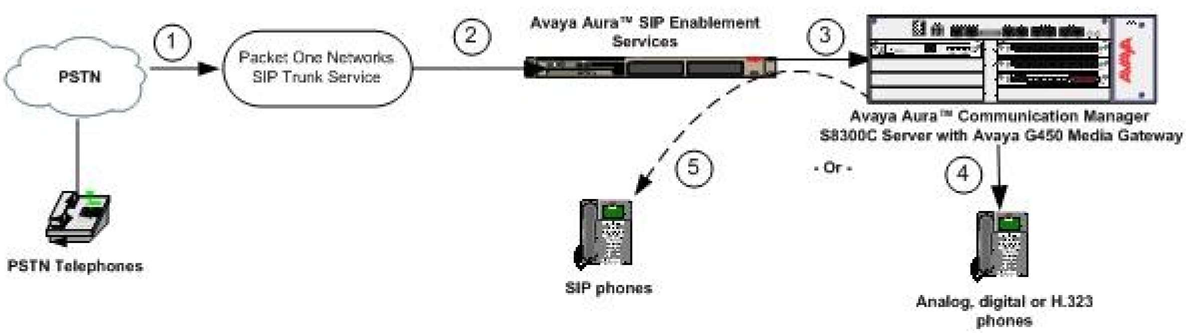

Figure 2: Incoming PSTN Calls to Communication Manager

Appendix A illustrates an example of a SIP INVITE message sent by Packet One Networks 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 Packet One Networks.

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

Or

2. A SIP telephone originates a call that is routed via Avaya Aura™ SIP Enablement Services (as shown by arrow 2) to Communication Manager.
3. The call request is handled by Avaya Aura™ Communication Manager where origination services and call routing are performed. Avaya Aura™ Communication Manager selects the SIP trunk and sends the SIP signaling messages to Avaya Aura™ SIP Enablement Services.
4. Avaya Aura™ SIP Enablement Services routes the call to Packet One Networks.
5. Packet One Networks completes the call to the PSTN.

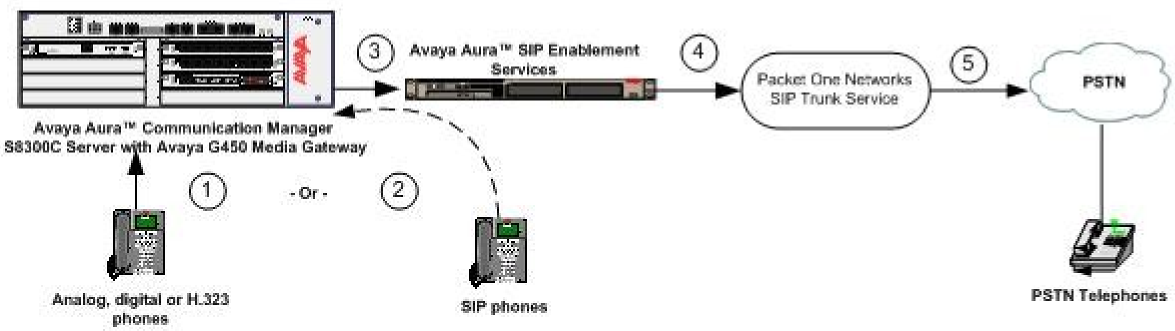

Figure 3: Outgoing Calls from Communication Manager to the PSTN

2. Equipment and Software Validated

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

Equipment	Software
S8300 Server with an Avaya G450 Media Gateway	Avaya Aura^{TM} Communication Manager R5.2 R015x.02.0.947.3 Update: Service Pack 3
Avaya Aura^{TM} SIP Enablement Services on S8500 Server	Avaya Aura^{TM} SIP Enablement Services SES05.2-02.0.947.3b Update: Service Pack 2
Avaya 9600 series IP Telephone	R3.0 – H.323
Avaya 9600 series IP Telephone	R2.4.2 - SIP
Avaya 2420 Digital Telephone	n/a
Analog Telephone	n/a
Comverse Session Border Controller	ALGSIP 4.20.51
Comverse CCS Softswitch	4.44.6
AudioCodes Mediant 2000 Gateway	5.40.037.004
SMCSS7	isupIP:2.6.18, sipITF:1.0.97, smcloop:1.2.31

Table 1: Equipment and Software Tested

Note: This solution will be compatible with other Avaya Server and Media Gateway platforms running similar versions of Communication Manager and SIP Enablement Services.

3. Configure Avaya Aura^{TM} Communication Manager

This section describes the steps for configuring Communication Manager for SIP Trunking. SIP trunks are established between Communication Manager and SIP Enablement Services (SES). These SIP trunks will carry SIP signaling associated with the Packet One Networks SIP Trunk Service as well as signaling associated with SIP endpoint devices. Avaya SIP telephones are configured as off-pbx stations (OPS) on Communication Manager. These SIP stations register with SIP Enablement Services but have calling privileges and features managed by Communication Manager. 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 **4.2** describing SIP endpoint administration may be omitted if SIP endpoints are not used. In the Avaya SIP architecture, SIP Enablement Services acts as a SIP proxy through which all incoming and outgoing SIP messages flow from and to the Packet One Networks SIP Trunk Service. There is no direct SIP signaling path between Packet One Networks and Communication Manager or Avaya SIP endpoints.

For incoming calls, the SIP Enablement Services uses address maps to direct the incoming SIP messages to the appropriate Communication Manager, as shown in **Section 4.1.5**. Once the message arrives at 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 Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects a SIP trunk, the SIP signaling is routed to the SIP Enablement Services. The SIP Enablement Services directs the outbound SIP messages to the Packet One network.

The dial plan for the configuration described in these Application Notes consists of 00+9-digit dialing for local and long-distance calls over the PSTN. Communication Manager routes all calls to the Packet One network using Automatic Route Selection (ARS). 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 S8300 Server and Avaya G450 Media Gateway is presumed to have been previously completed and is not discussed here.

3.1. SIP Trunk Configuration

The following procedures include:

- Confirm Necessary Optional Features
- Administer IP Nodes Names
- Administer IP Network Regions
- Administer IP Codec Sets
- Administer SIP Signaling Groups
- Administer SIP Trunk Groups
- Administer Calling Party Number Information
- Administer Automatic Route Selection for Outbound Calls
- Administer Incoming Digit Translation
- Save Avaya Aura™ Communication Manager Changes

3.1.1. Confirm Necessary Optional Features

Log in to the 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: 22                             RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 900   59
                                      Maximum Stations: 450   8
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 0     0
                    Maximum Off-PBX Telephones -   OPS: 450   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 Packet One network, SIP endpoints and any other SIP trunks used. Each Avaya SIP telephone on a 2-party call with Packet One 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 Packet One uses one SIP trunk.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 450   10
           Maximum Concurrently Registered IP Stations: 450   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 Stations: 100   0
                   Maximum Video Capable IP Softphones: 100   0
                       Maximum Administered SIP Trunks: 450   41
```

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.

3.1.2. Administer IP Node Names

The node names defined here will be used in other configuration screens to define a SIP signaling group between Communication Manager and SIP Enablement Services. In the **IP Node Names** form, assign the node name and IP address for the SIP Enablement Services Server (SES) at the enterprise site as shown in **Figure 6**. In this case, **SES** and **10.10.10.10** are the name and IP Address for the SIP Enablement Services. In other Avaya configurations such as an Avaya G650 Media Gateway with an Avaya S8720 or S8500 Server, the C-LAN interface address must be used as the SIP signaling interface to SIP Enablement Services, rather than the processor address (node name **procr**).

```
display node-names ip
                                   IP NODE NAMES
    Name              IP Address
default             0.0.0.0
SES                 10.10.10.10
procr               10.10.10.11
```

Figure 6: IP Nodes Names Form

3.1.3. Administer 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 Packet One Networks 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 or processor interface 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 Packet One 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 Packet One, 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 Packet One 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 SIP Enablement Services. In this configuration, the domain name is **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 VoIP resources. Packet One supports shuffling to direct **IP-IP Direct Audio** so these parameters can retain the **yes** default values, however they can be set to no if shuffling is not required.
- 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 Packet One.

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

Figure 7: IP Network Region 1 – Page 1

Navigate to **Page 3**. Whilst configuring ip-network region 1 the source region will be 1. 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 Packet One network, which will logically reside in network region 2.

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

 Source Region: 1     Inter Network Region Connection Management     I       M
                                                                     G  A    e
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    a
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    s
 1    1                                                               all
 2    2     y    NoLimit                                             n all
```

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 VoIP resources.
- 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 Packet One 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 Packet One also goes out the SIP trunk to Packet One.

```
change ip-network-region 2                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 2
Location:         Authoritative Domain:
    Name: Packet One
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**. Whilst configuring ip-network region 2 the source region will be 2. 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 Packet One network.

```
change ip-network-region 2                                      Page   3 of  19

 Source Region: 2     Inter Network Region Connection Management     I       M
                                                                     G  A    e
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    a
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    s
 1    2     y    NoLimit                                             n all
 2    2                                                                all
```

Figure 10: IP Network Region 1 – Page 3

3.1.4. Administer IP Codec Sets

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. 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 Packet One 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:
```

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 Packet One network, in order of preference. The settings of the **IP Codec Set** form are shown in **Figure 12.** G.729A is the codec to be used for connections to the Packet One 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 and G.729A.

```
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:
```

Figure 12: IP Codec Set 2 Form – Page 1

The Packet One network supports the T.38 fax protocol. If calls involving fax machines will be made using the Packet One network, it is necessary to configure the T.38 fax protocol by setting the **Fax Mode** to **t.38-standard** on **Page 2** of the codec set form as shown below.

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

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

Figure 13: IP Codec Set 2 Form – Page 2

3.1.5. Administer SIP 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 Packet One network. Another PSTN Inbound signaling group (and trunk group) will be used for inbound calls from the Packet One network. A third SIP OPS signaling group is defined for calls involving SIP telephones. Recall that SIP telephones register with SIP Enablement Services and leverage the calling privileges and features provided by Communication Manager. The configuration steps below show the configuration of these signaling groups.

Configure the PSTN Outbound **Signaling Group** using the **add signaling-group n** command, where **n** is an available signaling group, as 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 the processor interface (node name **procr**). 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 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 Packet One 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 Packet One network will be between region 1 and region 2.
- Enter the IP Address of the Packet One network element (provided by Packet One) 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 Packet One, 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**. Packet One supports the Avaya **Direct IP-IP Audio** feature. This feature can be disabled if desired, so that the media gateway handles all the voice data for outgoing calls.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833, as specified in reference **[7]**.
- 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
  IMS Enabled? n
     IP Video? n




   Near-end Node Name: procr                  Far-end Node Name: SES
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 2
Far-end Domain: 10.20.20.20

                                             Bypass If IP Threshold Exceeded? n

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

Figure 14: PSTN-Outbound Signaling Group Form

Configure the PSTN Inbound **Signaling Group** using the **add signaling-group n** command, where **n** is an available signaling group, as 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 the processor interface (node name **procr**). 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 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 Packet One 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 Packet One network to Avaya devices will be between region 2 and region 1.
- Leave the **Far-end Domain** field blank, allowing inbound PSTN calls from Packet One to be accepted using this signaling group.
- The **Direct IP-IP Audio Connections** field is set to **y**. Packet One supports the Avaya **Direct IP-IP Audio** feature. This feature can be disabled if desired, so that the media gateway handles all the voice data for incoming calls.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value **enables Com**munication Manager to send DTMF transmissions us**ing RFC 283**3, as specified in reference **[7]**.
- The default values for the other fields may be used.

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

 Group Number: 3              Group Type: sip
                        Transport Method: tls
  IMS Enabled? n
     IP Video? n


   Near-end Node Name: procr                  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
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                     Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? y            Alternate Route Timer(sec): 6
```

Figure 15: PSTN-Inbound Signaling Group Form

Configure the SIP OPS **Signaling Group** using the **add signaling-group n** command, where **n** is an available signaling group, as 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 the processor interface (node name **procr**). 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 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 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
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
                        Transport Method: tls
  IMS Enabled? n
     IP Video? n


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

                                             Bypass If IP Threshold Exceeded? n

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

Figure 16: SIP OPS Signaling Group Form

3.1.6. Administer SIP Trunk Groups

One trunk group will be associated with each of the signaling groups described in **Step 3.1.5.** Configure the PSTN Outbound **Trunk Group** form as shown in **Figure 17** using the **add trunk-group x** command, where **x** is an available trunk group. 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 Packet One network.

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

Group Number: 2                    Group Type: sip           CDR Reports: y
  Group Name: PSTN Outbound                COR: 1        TN: 1        TAC: 302
   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 **1200**. If the default value of 600 is retained in this field, each outbound SIP call to Packet One will require additional, avoidable SIP messaging that can perceptibly delay call establishment. With this value set to 1200, the initial SIP INVITE message from Avaya to Packet One will contain a value the Packet One 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: auto
                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 1200
```

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
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. Set the **Send Diversion Header** field to **y**. Diversion headers are required by Packet One when a call is forwarded from Communication Manager to an external PSTN number. As shown in **Figure 20**, leave the **Telephone Event Payload Type** (associated with DTMF transmission using RFC 2833) to the default value of "blank". The Packet One platforms used for the compliance test are capable of negotiating to an alternate telephone event payload type offered by Avaya. Packet One may also recommend a value to be used 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

                     Send Diversion Header? y
                   Support Request History? y
             Telephone Event Payload Type:
```

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 y** command, where **y** is an available trunk group. 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 Packet One network.

```
add trunk-group 3                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 3                    Group Type: sip           CDR Reports: y
  Group Name: PSTN inbound                COR: 1        TN: 1        TAC: 303
   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 **1200**. If the default value of 600 is retained in this field, inbound SIP calls from Packet One 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 Packet One trunk calls to IP Telephones.

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

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                                Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 1200
```

Figure 22: Inbound PSTN Trunk Group Form – Page 2

Navigate to **Page 3** of the **Trunk Group** form. As shown in **Figure 23**, set the **Numbering Format** field to **public**. Since this trunk group is used for incoming PSTN trunk calls, optionally, the 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 Packet One-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 23: Inbound PSTN Trunk Group Form – Page 3

Navigate to **Page 4** of the **Trunk Group** form. As shown in **Figure 24**, set the **Telephone Event Payload Type** to the default value of **blank**. Packet One may also recommend a value to be used 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

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

Figure 24: Inbound PSTN Trunk Group Form – Page 4

Configure the SIP OPS **Trunk Group** form as shown in **Figure 25** using the **add trunk-group n** command, where **n** is an available trunk group. 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 SIP Enablement Services. For example, an outbound call from a SIP Telephone to Packet One will use one trunk member from trunk group 1 and one trunk member from trunk group 2. An incoming call from Packet One 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                  COR: 1         TN: 1        TAC: 301
   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: 30
```

Figure 25: SIP OPS PSTN Trunk Group Form – Page 1

Navigate to **Page 3** of the **Trunk Group** form. As shown in **Figure 26**, set the **Numbering Format** field to **public**.

```
add 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 26: SIP OPS Trunk Group Form – Page 3

3.1.7. Administer Calling Party Number Information

Use the **change public-unknown-numbering** command shown in **Figure 27** to configure Communication Manager to send the calling party number. In the sample configuration, all stations with a 4-digit extension beginning with 1 will send the calling party number 100015-xxxx to Packet One. 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: 2
 4  1               2         100015        10          Maximum Entries: 240
```

Figure 27: Format For Calling Party Number

3.1.8. Administer 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 Packet One. 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 28** 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 4 digit extensions with first digit 1 can also be observed.

```
change dialplan analysis                                             Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all          Percent Full:    0

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     1            4    ext
     2            5    ext
     3            3    dac
     4            5    ext
     6            5    ext
     7            5    ext
     8            5    ext
     9            1    fac
     *            3    fac
     #            3    fac
```

Figure 28: Dialplan Analysis Form

Use the **change feature-access-codes** command to configure or observe 9 as the ARS access code, as shown in **Figure 29**.

```
change feature-access-codes                                          Page   1 of   9
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: *01
         Abbreviated Dialing List2 Access Code: *02
         Abbreviated Dialing List3 Access Code: *03
Abbreviated Dial - Prgm Group List Access Code: *04
                      Announcement Access Code: *05
                       Answer Back Access Code: *06
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: *00
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: *07    Deactivation: #07
Call Forwarding Activation Busy/DA: *08    All: *09    Deactivation: #09
   Call Forwarding Enhanced Status: *10    Act: *11    Deactivation: #11
                         Call Park Access Code: *12
                       Call Pickup Access Code: *13
CAS Remote Hold/Answer Hold-Unhold Access Code: *14
                  CDR Account Code Access Code: *15
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation: *16    Deactivation: #16
                   Contact Closure   Open Code: *17      Close Code: #17
```

Figure 29: 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, all outgoing calls to Packet One were prefixed by the digits “00”.

Figure 30 shows the **ars analysis** configuration used during testing. Calls are sent to Route Pattern 2, which will contain the Outbound PSTN SIP Trunk Group to Packet One.

```
change ars analysis 00                                               Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    0

          Dialed            Total      Route     Call   Node  ANI
          String           Min  Max  Pattern    Type   Num   Reqd
    00                     8    12    2         pubu         n
```

Figure 30: ARS Analysis Form

Use the **change route-pattern** command to add the SIP trunk group to the route pattern that ARS selects, as shown in **Figure 31**. In this configuration, route pattern 2 is used to route calls to trunk group 2.

```
add route-pattern 2                                         Page   1 of   3
                    Pattern Number: 2   Pattern Name: To Packet One
                             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                                                           n   user
 2:                                                                  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                                          none
 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 31: Route-Pattern Containing Outbound PSTN SIP Trunk Group

3.1.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DID calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from Packet One 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 Packet One correlate to the internal extensions assigned within Communication Manager. The first entry in **Figure 32** below translates incoming DID numbers in the range 100015-1xxx to the corresponding the 4 digit extension 1xxx by deleting the leading 6 digits.

```
change inc-call-handling-trmt trunk-group 3                    Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    10 1000151          6
```

Figure 32: Incoming Call Handling Treatment

Save Communication Manager changes by enter **save translation** to make them 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 Packet One SIP Trunk Service. The following procedures include:

- Assigning a Station
- Administer Off PBX Station Mapping
- Assigning Additional Stations
- Save Communication Manager Changes

3.2.1. Assigning a Station

Assign a station as shown in **Figure 33**. This example uses an Avaya one-X 9640 Deskphone. Using the **add station** command from the SAT:

- Set the station **Type** to the value **9640**.
- 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 SIP Enablement Services.

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

```
add station 1007                                             Page   1 of   6
                                     STATION

Extension: 1007                         Lock Messages? n             BCC: 0
     Type: 9640                         Security Code:                TN: 1
     Port: S00017                     Coverage Path 1:               COR: 1
     Name: Jon 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: 1007
            Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english                 Button Modules: 0
```

Figure 33: 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 34**.

```
add station 1007                                              Page   4 of   5

BUTTON ASSIGNMENTS
 1: call-appr                             5: cpn-unblk
 2: call-appr                             6: call-pkup
 3: call-appr                             7:
 4: cpn-blk                               8:
```

Figure 34: Station Administration – Page 4

3.2.2. Administer 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 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 35**:

- 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 SIP Enablement Services match the extensions of the corresponding stations on 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 1007                   Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number     Trunk       Config  Dual
 Extension                 Prefix                      Selection   Set     Mode
 1007             OPS           -     1007              1          1
```

Figure 35: 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 1007                   Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged         Location
 Extension     Limit      Mode         Allowed     Calls
 1007           3       both           all         none
```

Figure 36: Stations with Off-PBX Telephone Integration – Page 2

3.2.3. Assigning Additional Stations

Steps 3.2.1 and **3.2.2** can be repeated for each SIP phone to be added.

3.2.4. Save Avaya Aura^{TM} Communication Manager Changes

Enter **save translation** to make the changes permanent.

3.3. Configuration of Non-G.729A SIP Endpoints

The Packet One 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 Packet One SIP Trunk Service. In the compliance testing, calls involving Avaya 4600-Series SIP Telephones successfully communicated using G.711MU to the Avaya G450 Media Gateway, which in turn presented G.729A on the leg of the connection facing the Packet One network.

4. Configure Avaya Aura™ SIP Enablement Services

This section covers the administration of SIP Enablement Services (SES). SIP Enablement Services is configured via an Internet browser using the Administration web interface. It is assumed that SIP Enablement Services software and the license file have already been installed. For additional information on installation tasks, refer to **[4]**. This section is divided into two parts: **Section 4.1** provides the steps necessary to configure a SIP trunk to Packet One. **Section 4.2** provides the steps necessary to complete the administration for optional SIP endpoints.

4.1. SIP Trunking to Packet One

The following procedures include:

- Logging onto SIP Enablement Services
- Verifying System Properties
- Verifying SIP Enablement Services Host properties
- Add Communication Manager Server to SIP Enablement Services
- Administer Address Maps to Communication Manager
- Administer Address Maps to Packet One
- Administer Trusted Host(s)

4.1.1. Logging onto Avaya Aura™ 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 SIP Enablement Services server. Log in with the appropriate credentials and then select the **Administration** link and then **SIP Enablement Services** from the main screen as shown in **Figure 37**.

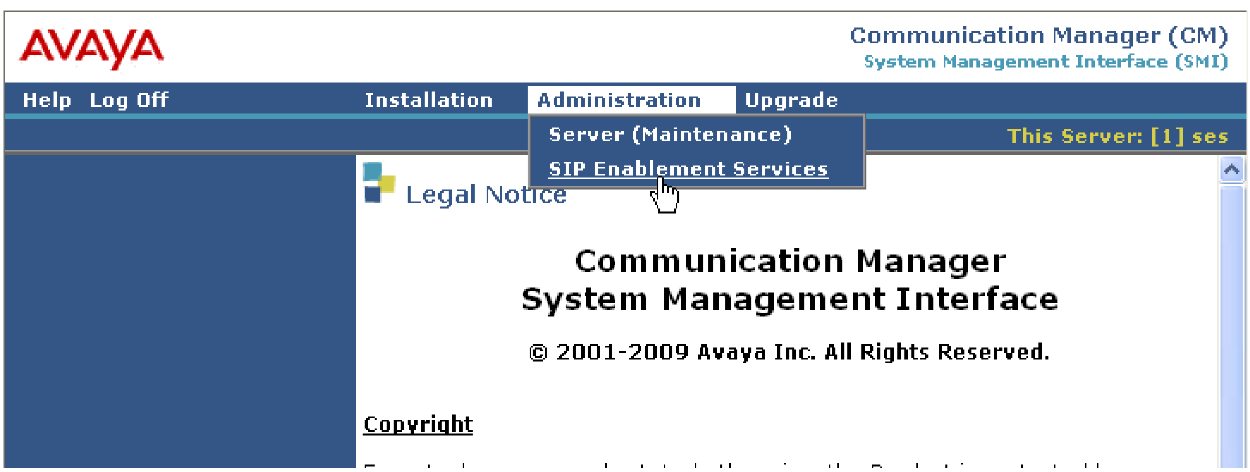

Figure 37 - SIP Enablement Services Main Screen

The SIP Enablement Services administration home screen shown in **Figure 38** will be displayed.

Figure 38: SIP Enablement Services Administration Home Page

4.1.2. Verifying System Properties

From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. This screen displays the SIP Enablement Services version and network properties configured during the installation process. In the **View System Properties** screen, verify the **SIP Domain** name assigned to SIP Enablement Services. This domain should match the domain configured in Communication Manager for the network region for local users (**Figure 7**) and the SIP signaling group to SES for SIP OPS Telephones (**Figure 16**).

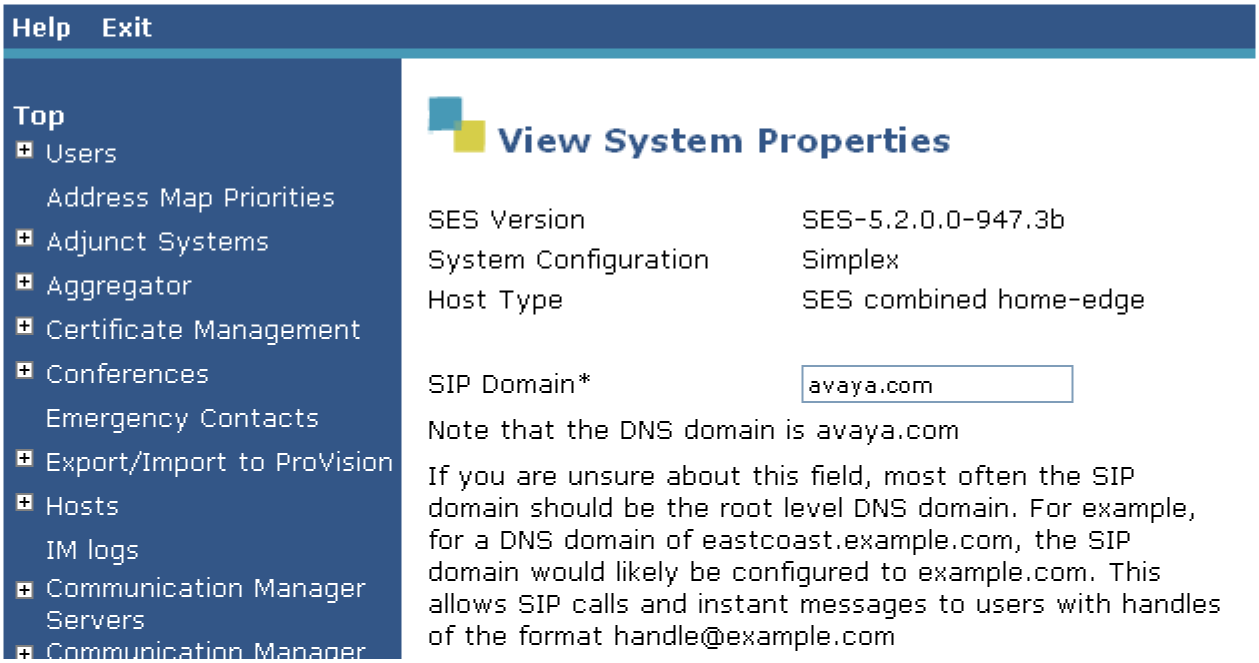

Figure 39: System Properties Showing SIP Domain

4.1.3. Verifying SES Host Properties

Verify the SIP Enablement Services Host information using the **Edit Host** page. In these Application Notes, the SIP Enablement Services **Host Type** is a combined **home/edge**. This means that both the Packet One SIP Trunk Service and Communication Manager are contacting the same SIP Enablement Services. Display the **Edit Host** page (**Figure 40)** 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 40**:

- Verify the **Host IP Address** of this combined SIP Enablement Services 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.

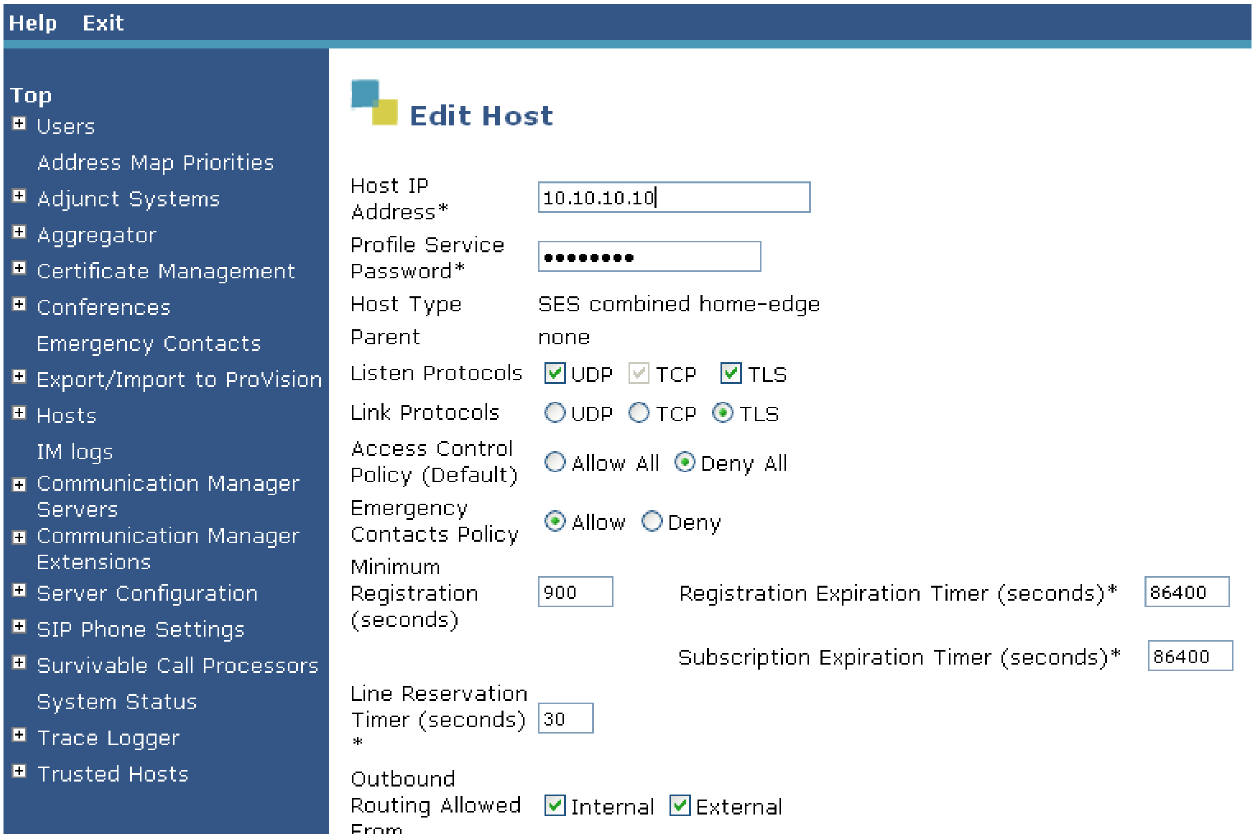

Figure 40: Edit Host

4.1.4. Add Avaya Aura™ Communication Manager Server

Expand the **Communication Manager Servers** option in the Administration web interface, and select **Add**. This step will create the SIP Enablement Services side of the SIP trunk previously created in Communication Manager. In the **Add Communication Manager Server Interface** screen, enter a descriptive name in the **Communication Manager Server Interface Name** field (e.g., **S8300-G450**). The IP Address of the single Home/Edge SIP Enablement Services Server is automatically entered in the **Host** field. Select TLS (Transport Layer Security) for the **SIP Trunk Link Type**. Enter the IP address of the processor interface used in the definition of the SIP signaling group to SIP Enablement Services (**Figure 16**) in the **SIP Trunk IP Address** field. In alternate configurations such as those using the Avaya S8720 or S8500 Server, this may be the IP address of the C-LAN board. Scroll to the bottom, and click **Add** (not shown).

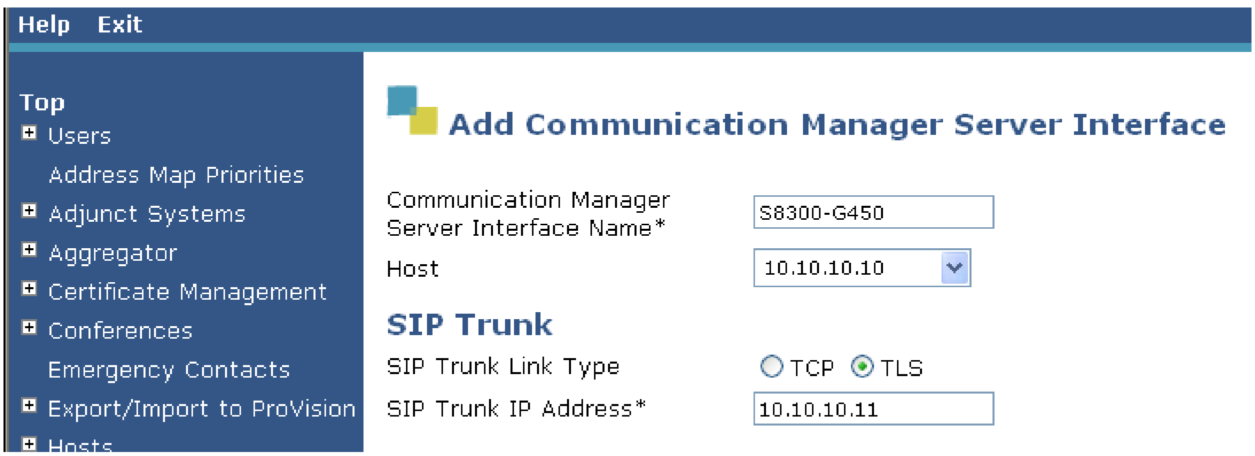

Figure 41: Add Communication Manager Server Interface

4.1.5. Administer Address Maps to Avaya Aura™ Communication Manager

Incoming calls arriving at SIP Enablement Services are routed to the appropriate Communication Manager for termination services. This routing is specified in a Communication Manager Address Map configured on 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 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 Communication Manager systems supported by the same SIP Enablement Services.

For the compliance testing, 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 Packet One would be:

sip:1000151003@10.10.10.10;user=phone;

The user portion in this case is the 10 digit DID number "1000151003". One or more address maps can be created to match the DID numbers assigned to the customer by Packet One. SIP Enablement Services will forward the messages based on the matching patterns to the appropriate *domain* IP address or name, in this example the *domain* IP address will be the Avaya S8300 Server processor interface.

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

- Select **Communication Manager Servers** in the left pane of the Administration web interface.
- Click **List Communication Manager Servers**.
- 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 42:**

- Enter a descriptive name in the **Name** field
- Enter the regular expression to be used for the pattern matching in the **Pattern** field. In this configuration, the DID number range provided by Packet One is 100015-1xxx. A pattern specification (without the double quotes) for these DID numbers is: **^sip:1000151[0-9]{3}**. URIs beginning with **sip:1000151** followed by 3 digits from 0 to 9. **Appendix B** provides an overview of the syntax for address map patterns.
- Leave the **Replace URI** check box selected
- Click the **Add** button once the form is completed

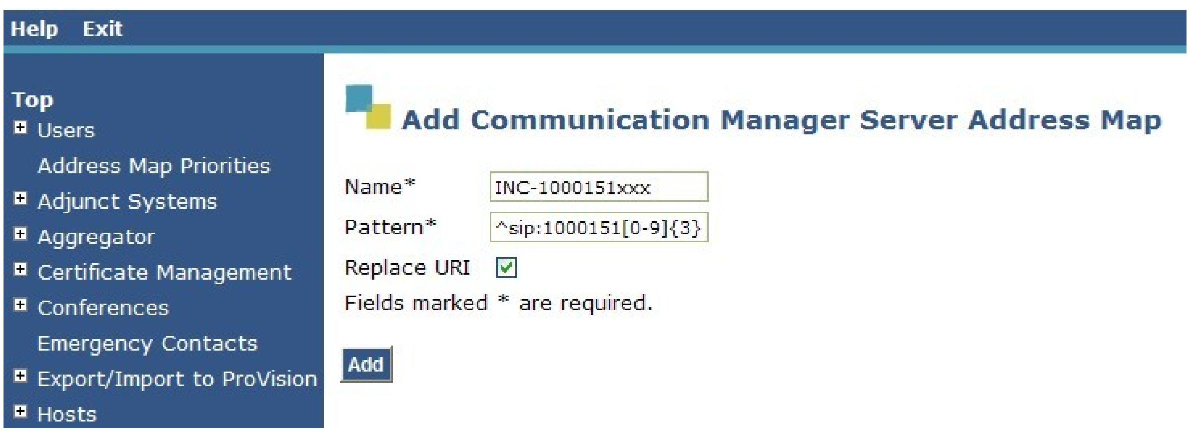

Figure 42: Add Communication Manager Address Map

After adding the address map, the **List Communication Manager Server Address Map** screen will appear, as shown in **Figure 43**.

Figure 43: List Communication Manager 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 42**, the following contact was created:

sip:$(user)@10.10.10.11:5061;transport=tls

The contact specifies the IP address of the S8300 Server processor interface 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).

4.1.6. Administer Address Maps to Packet One

Outbound PSTN calls are directed by Communication Manager automatic route selection (ARS) according to the customer's network design guidelines. These guidelines determine what types of outgoing calls should be sent to the Packet One SIP Trunk Service. The ARS routing decisions (for trunk group selection) will be customer specific and are beyond the scope of these Application Notes. SIP signaling messages for outbound calls sent to the SIP trunk are then routed to the Packet One SIP proxy using Host Address Maps within SIP Enablement Services. These Host Address Maps use pattern matching on the SIP URI to direct messages to the corresponding contact address (e.g., the Packet One SIP signaling proxy). In this configuration, the SIP Enablement Services routing rule for the SIP trunk group will be to send all outbound PSTN traffic to Packet One SIP Trunk Service. To perform this, several dialing patterns will be created in the SIP Enablement Services. An example is the pattern of **^sip:001[0-9]{9}** will match on all calls having digits beginning with 001.

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.

- To configure an outbound **Host Address Map**: Select **Hosts** in the left pane of the Administration web interface.
- Click **List Hosts**.
- Click on the **Map** link associated with the appropriate host.
- Click on the **Add Map In New Group** link.
- In the screen shown in **Figure 44:** Enter a descriptive name in the **Name** field.
- Specify an appropriate pattern for the call type. In this example, the pattern used is **^sip:001[0-9]{9}**.
- Leave the **Replace URI** checkbox selected.
- Click the **Add** button once the form is completed.

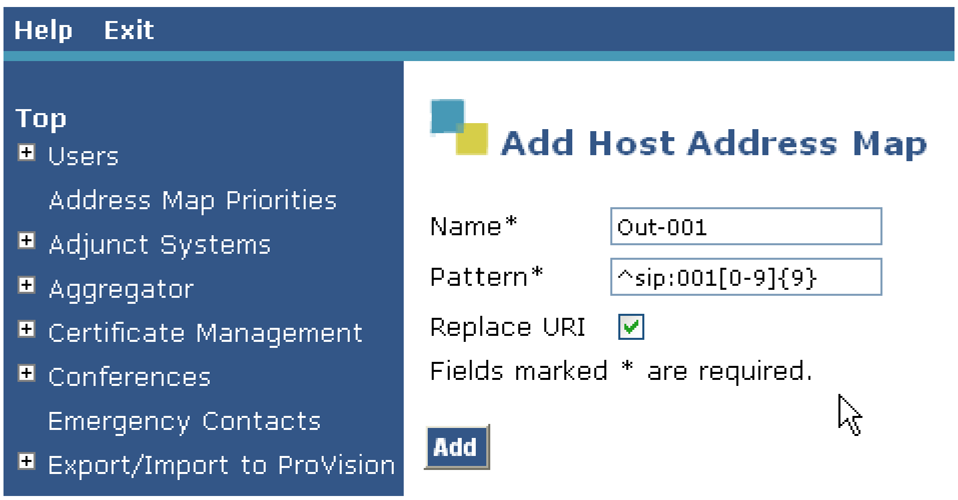

Figure 44: Add Host Address Map

The next step is to enter the contact address for the Packet One SIP Proxy. In this example, an IP address is used to identify Packet One SIP Proxy. The customer's specific information will be provided by Packet One. To enter the Packet One SIP Proxy information: Display the **List Host Address Map** screen. Click on the **Add Another Contact** link associated with the address map added in **Figure 44** 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)@10.20.20.20:5060;transport=tcp

The user part in the original request URI is inserted in place of the **$(user)** string before the message is sent to Packet One. 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 45**.

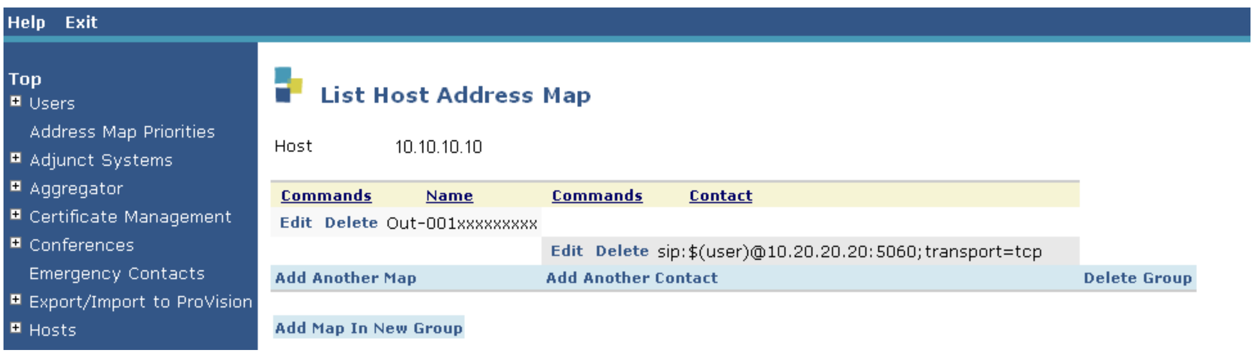

Figure 45: List Host Address Map

4.1.7. Administer Trusted Host(s)

The IP addresses provided by Packet One for SIP network elements must be added as trusted hosts to the SIP Enablement Services. For a trusted host, SIP Enablement Services will not issue SIP authentication challenges for incoming requests from the designated IP address. If multiple SIP proxies are used in the Packet One network to route calls to the SIP Enablement Services in the enterprise, the IP address of each must be added as a trusted host.

Expand **Trusted Hosts** from the lower left of the SIP Enablement Services Administration page (shown in **Figure 38**). Click **Add**. In the **Add Trusted Host** screen shown in **Figure 46**, enter the IP Address provided by Packet One for the Packet One 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 SIP Enablement Services for which the trust relationship must exist. Click **Add**.

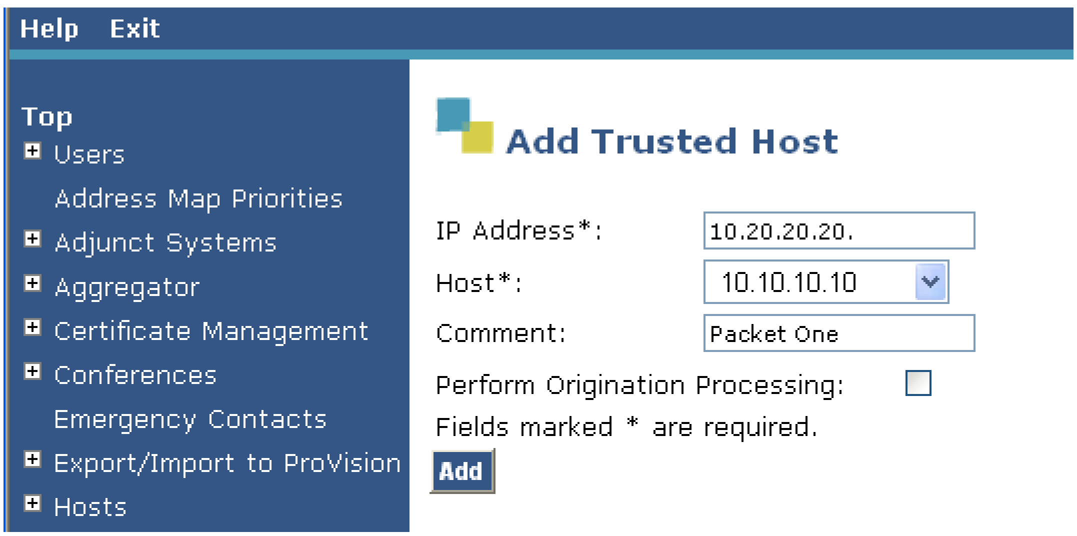

Figure 46: Adding a Trusted Host

A screen like **Figure 47** will appear. Click **Continue**.

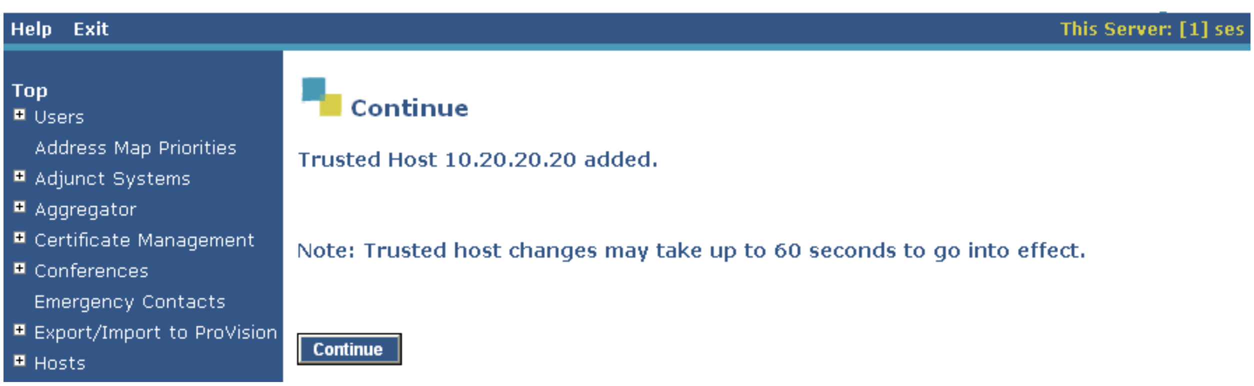

Figure 47: Continue Adding a Trusted Host

4.2. Configuration for Optional SIP Telephones

This section provides basic instructions for completing the SIP Enablement Services administration necessary to support optional Avaya SIP telephones. The following procedures include:

- Adding a SIP User
- Administering Additional SIP Telephones

4.2.1. Adding a SIP User

In SIP Enablement Services Administration, expand **Users**. Click **Add**. In the **Add User** screen shown in **Figure 48**:

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

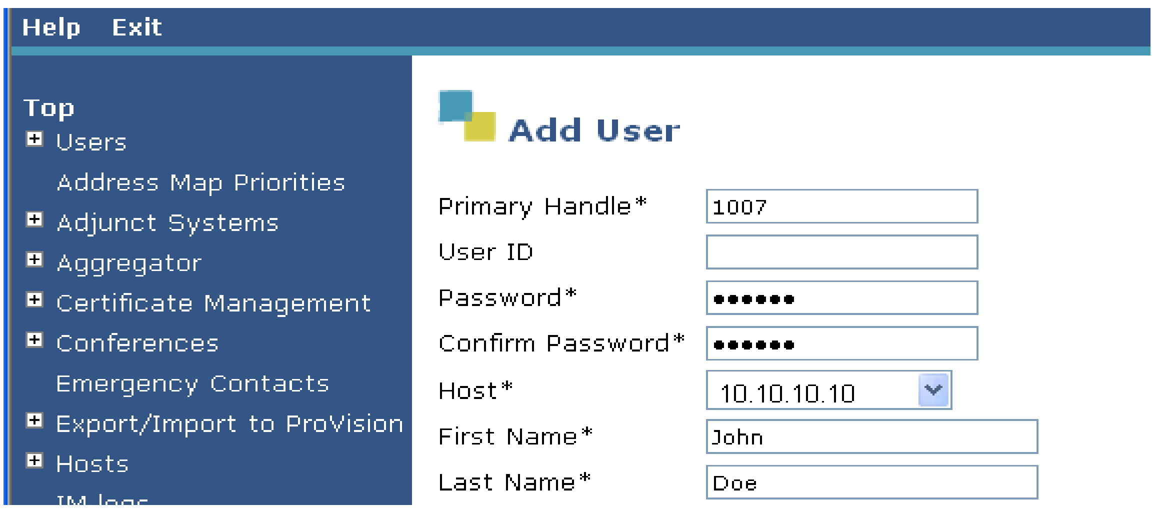

Figure 48: Add User – User Information

Scroll to the bottom of the **Add User** page, and select the **Add Communication Manager Extension** checkbox as shown in **Figure 49**. Click **Add**.

Figure 49: Add User – Add Communication Manager Extension Area

Press **Continue** at the confirmation screen and the **Add Communication Manager Extension** screen will appear. Shown in **Figure 50**:

- Enter the **Extension** configured on Communication Manager, configured in **Figure 33**.
- From the drop-down, select the **Communication Manager Server** associated with this extension.
- Click **Add**.

Figure 50: Add Communication Manager Extension

4.2.2. Administering Additional SIP Telephones

Repeat all the steps in **Section 4.2.1** for each additional SIP user.

5. Packet One Services Configuration

To use Packet One SIP Trunk Service, a customer must request service from Packet One using their sales processes.

6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunk interoperability between the Packet One 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 SIP Trunk Service provided by Packet One. The compliance test included the following:

- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by Packet One. Incoming PSTN calls were made to H.323, digital and analog telephones at the enterprise.

- Outgoing calls from the enterprise site were completed via Packet One to PSTN destinations. Outgoing calls from the enterprise to the PSTN were made from H.323, digital and analog telephones.
- Calls using G.729A, G.711MU, and G.711A codecs.
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using the T.38 mode.
- 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, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction.
- Direct IP-to-IP media (also known as “shuffling”) with SIP and H.323 telephones.
- Voicemail coverage and retrieval for endpoints at the enterprise site.
- Packet One network-based maintenance via periodic transmission of SIP OPTIONS messages by Packet One requiring Avaya response.

6.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the Packet One SIP Trunk Service.

7. Verification Steps

This section provides verification steps that may be performed in the field to verify that the H.323, digital and analog endpoints can place outbound and receive inbound PSTN calls using the Packet One 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. Conclusion

These Application Notes describe the configuration steps enabling customers using Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services to connect to the PSTN via the Packet One SIP Trunk Service.

9. References

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

[1] *SIP Support in Avaya Communication Manager Running on Avaya Servers,* May 2009, Document Number 555-245-206.

[2] *Administering Avaya Aura™ Communication Manager,* May 2009, Document Number 03-300509

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

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

[5] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Installation and Maintenance Guide Release 3.0,* Feb-2009, Document Number 16-300694

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

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

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 Packet One to the Avaya SES at the enterprise site. The call is from a PSTN telephone user to the Packet One-provided DID 1000151002.

Sample SIP INVITE Message from Packet One to Avaya SIP Enablement Services:

INVITE sip:1000151002@avaya.com;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.20.20.20:5060;maddr=10.20.20.20;branch=z9hG4bK1c1d106b839435f4c6b32295279d3b15
Record-Route: <sip:1000151002@10.20.20.20:5060;lr>
From: <sip:1100@10.20.20.20;user=phone>;tag=z07m5db13cb-cbs1542642092-o-53-187955706
To: <sip:1000151002@avaya.com;user=phone>
Call-ID: 98a021c8c4ba11de95b5000c299366a20114a8c0@10.20.20.20
CSeq: 1 INVITE
Contact: <sip:_53v1542642092@10.20.20.20:5060>
P-charging-vector: icid-value=12be2b7bc-1
Max-forwards: 58
P-asserted-identity: <sip:1100@10.20.20.29;user=phone>
Privacy: none
X-ncx-service-info: ncx-info=0x407100190001c00f653136343a31303030363231313030019801000100
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, REFER, PRACK, INFO, MESSAGE, SUBSCRIBE, NOTIFY, UPDATE
Content-Type: application/sdp
Content-Length: 851

v=0
o=SVI-SDP 1 0 IN IP4 10.165.106.125
s=-
c=IN IP4 10.20.20.21
t=0 0
m=audio 4040 RTP/AVP 118 0 8 97 98 2 99 105 106 107 108 109 110 111 112 113 4 80 18 3 116 96 13 120
a=rtpmap:118 G729B/8000/1
a=fmtp:118 annexb=yes
a=rtpmap:97 G726-16/8000/1
a=rtpmap:98 G726-24/8000/1
a=rtpmap:99 G726-40/8000/1
a=rtpmap:105 X-G727-16/8000/1
a=rtpmap:106 X-G727-24-16/8000/1
a=rtpmap:107 X-G727-24/8000/1
a=rtpmap:108 X-G727-32-16/8000/1
a=rtpmap:109 X-G727-32-24/8000/1
a=rtpmap:110 X-G727-32/8000/1
a=rtpmap:111 X-G727-40-16/8000/1
a=rtpmap:112 X-G727-40-24/8000/1
a=rtpmap:113 X-G727-40-32/8000/1
a=rtpmap:4 G723/8000/1
a=fmtp:4 bitrate=6.3;annexa=yes
a=rtpmap:80 G723/8000/1
a=fmtp:80 bitrate=5.3;annexa=yes
a=fmtp:18 annexb=yes
a=rtpmap:116 X-CCD/8000/1
a=rtpmap:96 telephone-event/8000

a=fmtp:96 0-15
a=rtpmap:120 no-op/8000

Sample SIP INVITE Message from Avaya SIP Enablement Services to Packet One:
This trace corresponds to the initial INVITE for an outbound call from a H.323 telephone with extension 1004 and name “H.323 1” to PSTN destination 001000621100. The codec requested for the call is G.729. 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 10.10.10.12, which is the IP Address of the G450 Media Gateway.

```
INVITE sip:001000621100@10.20.20.20 SIP/2.0
Accept-Language: en
Call-ID: 0c612bef7dbde12c84aefaae700
CSeq: 1 INVITE
From: "H.323 1" <sip:1000151004@avaya.com>;tag=0c612bef7dbde12b84aefaae700
Record-Route: <sip:10.10.10.10:5060;lr>,<sip:
10.10.10.11:5061;lr;transport=tls>
To: "001000621100" <sip:001000621100@10.20.20.20>
Via: SIP/2.0/UDP
10.10.10.10:5060;branch=z9hG4bK030303363636636363632c1c.0,SIP/2.0/TLS10.10.10.1
1;psrrposn=2;received=10.10.10.11;branch=z9hG4bK0c612bef7dbde12d84aefaae700
Content-Length: 192
Content-Type: application/sdp
Contact: "H.323 1" <sip:1000151004@10.10.10.11:5061;transport=tls>
Max-Forwards: 70
User-Agent: Avaya CM/R015x.02.0.947.3
Allow:
INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS,INFO,PUBLISH
Supported: timer,replaces,join,histinfo,100rel
Alert-Info: <cid:internal@10.20.20.20>;avaya-cm-alert-type=internal
Min-SE: 2400
Session-Expires: 2400;refresher=uac
P-Asserted-Identity: "H.323 1" <sip:1000151004@avaya.com:5061>
History-Info: <sip:001000621100@10.20.20.20>;index=1,"001000621100"
<sip:001000621100@10.20.20.20>;index=1.1

v=0
o=- 1 1 IN IP4 10.10.10.11
s=-
c=IN IP4 10.10.10.12
b=AS:64
t=0 0
m=audio 2052 RTP/AVP 18 127
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
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, 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.