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

Abstract

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

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

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

These Application Notes describe the steps for configuring SIP trunking between the PAETEC Communications IPATH Service and an Avaya SIP Telephony solution consisting of Avaya SIP Enablement Services, Avaya Communication Manager and various Avaya telephony endpoints. These endpoints included IP telephones (using SIP and H.323 protocols), traditional analog and digital phones and the Avaya one-X Desktop Edition running on a Microsoft Windows PC.

PAETEC Communications currently serves customers in 29 markets. An illustration of PAETEC Communications network coverage is provided in Figure 1. An updated interactive version of this map may be found online at [http://www.paetec.com/2_1/2_1_5_2.html](http://www.paetec.com/2_1/2_1_5_2.html).

![Figure 1 – PAETEC Communications Market Coverage](image)
These Application Notes describe the configuration of SIP trunking to access the PAETEC Communications IPATH Service. Access to the PAETEC Communications service supports the following capabilities:

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

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

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

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

- An Avaya S8710 Media Server with an Avaya G650 Media Gateway. The S8710 served as the host processor for Avaya Communication Manager.
- Avaya SIP Enablement Services (SES) software operating on an Avaya S8500B server platform.
- Avaya 4600 Series IP telephones (configured to use either the SIP or H.323 protocol).
- Avaya 6400 Series digital and 6200 Series analog telephones.

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

Note also that security devices, such as firewall and network address translation (NAT) devices, are not included in this configuration. These Application Notes focused on SIP trunking interoperability. However, it is recommended that enterprise customers deploy security devices in a production environment.
Figure 2: Avaya SIP Telephony Network using PAETEC Communications IPATH Service

1.1. Call Flows

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

The first call scenario illustrated in Figure 3 is a PSTN call to the enterprise site terminating on a typical analog telephone supported by Avaya Communication Manager.

1. A user on the PSTN dials a PAETEC Communications provided DID number assigned to an Avaya Communication Manager telephone at the enterprise site. The PSTN routes the call to the PAETEC Communications network (as the local service provider) who routes the DID number to the assigned customer.

2. Based on the DID number, PAETEC Communications offers the call to Avaya SES using SIP signaling messages sent over the converged access facility. Note that the assignment of the DID number and the address of the Avaya SES server was previously established during the ordering and provisioning of the service.
3. Avaya SES routes the call to the Avaya S8710 Media Server running Avaya Communication Manager over a SIP trunk.

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

- or –

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

Figure 3: Incoming PSTN Calls to Avaya Communication Manager

Appendix A illustrates an example of a SIP INVITE message sent by PAETEC Communications for an incoming DID call.
The second call scenario illustrated in Figure 4 is an outgoing call from an Avaya telephone at the enterprise site to the PSTN via the SIP trunk to PAETEC Communications.

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

   - or -

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

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

3. Avaya SIP Enablement Services routes the call to PAETEC Communications.

4. PAETEC Communications completes the call to the PSTN.

![Figure 4: Outgoing Calls from Avaya Communication Manager to the PSTN](image)
2. Equipment and Software Validated
The following equipment and software were used for the sample configuration provided:

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

<table>
<thead>
<tr>
<th>PAETEC Communications IPATH Service Components</th>
<th>Version</th>
</tr>
</thead>
<tbody>
<tr>
<td>Lucent Distributed Network Controller/Gateway</td>
<td>5.1.0.3 SP 16</td>
</tr>
<tr>
<td>Acme Packet Net-Net Session Border Controller</td>
<td>2.1.0 P41</td>
</tr>
<tr>
<td>Broadworks Hosted Platform</td>
<td>12.0</td>
</tr>
</tbody>
</table>

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

3. Configure Avaya Communication Manager
This section describes the steps for configuring a SIP trunk on Avaya Communication Manager. The SIP trunk is established between Avaya Communication Manager and Avaya SIP Enablement Services server. This trunk will carry the SIP signaling sent to the PAETEC Communications IPATH Service.

This SIP trunk also provides the trunking for SIP endpoint devices such as Avaya 4600 SIP telephones and Avaya one-X Desktop Edition using Avaya Communication Manager in the recommended Off-PBX Station (OPS) configuration. Avaya SIP telephones are configured as OPS stations on Avaya Communication Manager. OPS SIP stations register with Avaya SES but have calling privileges and features managed by Avaya Communication Manager. Avaya Communication Manager acts as a back-to-back SIP user agent when a SIP phone places or receives a call over a SIP trunk to a service provider.

Note the use of SIP endpoints is optional. The steps discussed in Sections 3.2 and 4.2 describing SIP endpoints administration may be omitted if SIP endpoints are not used.
In the Avaya SIP architecture, the Avaya SES acts as a SIP proxy through which all incoming and outgoing SIP messages flow to PAETEC Communications. There is no direct SIP signaling path between PAETEC Communications and Avaya Communication Manager or Avaya SIP endpoints.

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

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

The dial plan for the configuration described in these Application Notes consists of 10-digit dialing for local and long-distance calls over the PSTN. In addition, Operator calls (0), Directory Assistance calls (411) and International calls (011+Country Code) were also supported. Avaya Communication Manager routes all calls using Automatic Route Selection (ARS), except for intra-switch calls. The configuration of ARS is beyond the scope of these Application Notes and the reader should refer to [1] and [2] for additional information.

The Avaya Communication Manager configuration was performed using the System Access Terminal (SAT). The general installation of the S8710 media server, G650 Media Gateway and circuit packs such as the CLAN is presumed to have been previously completed and is not discussed here.

### 3.1. SIP Trunk Configuration

#### Step 1: Confirm Necessary Optional Features

Using the SAT, verify that there exists sufficient SIP Trunks and Off-PBX Telephones capacities by displaying the System-Parameters Customer-Options form shown in Figure 5. The license file installed on the system controls the maximum permitted. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

On Page 1 of the System-Parameters Customer-Options form, verify that the number of OPS stations available is sufficient for the number of SIP telephones to be assigned. Maximum Off-PBX Telephones indicates the maximum number available in the system and is controlled by the license file. The USED column indicates the number of telephones licenses currently assigned. The difference between the two represents the additional number of SIP endpoints that can be added.
Figure 5: System-Parameters Customer-Options Form – Page 1

On Page 2, verify that the number of SIP trunks supported by the system is sufficient for the combination of trunks to the PAETEC Communications network, SIP endpoints and any other SIP trunks used. Note that calls from non-SIP endpoints to PAETEC Communications will use one SIP trunk for the duration of the call. Each SIP OPS telephone on a call with PAETEC Communications will require two SIP trunks for the duration of the call.

Figure 6: System-Parameters Customer-Options Form – Page 2

Step 2: Assign Node Names

In the IP Node Names form, assign the node name and IP address for Avaya SIP Enablement Services at the enterprise site. In this case “SES” and “10.1.1.86” are being used, respectively. The SES node name will be used throughout the other configuration screens of Avaya Communication Manager.
In this example “CLAN” and “10.1.1.84” are the name and IP address assigned to the TN799DP CLAN card. The CLAN entry was previously created during the installation of the system. Note, in smaller media gateways such as an Avaya G350, the S8300 processor address (procr) is used as the SIP signaling interface instead of the CLAN interface.

<table>
<thead>
<tr>
<th>Name</th>
<th>IP Address</th>
<th>Name</th>
<th>IP Address</th>
</tr>
</thead>
<tbody>
<tr>
<td>CLAN</td>
<td>10.1.1.84</td>
<td>default</td>
<td>. . .</td>
</tr>
<tr>
<td>default</td>
<td>10.1.1.84</td>
<td>ipsi</td>
<td>. . .</td>
</tr>
<tr>
<td>ipsi</td>
<td>10.1.1.84</td>
<td>medpro-hw11</td>
<td>. . .</td>
</tr>
<tr>
<td>medpro-hw11</td>
<td>10.1.1.84</td>
<td>procr</td>
<td>. . .</td>
</tr>
<tr>
<td>procr</td>
<td>. . .</td>
<td>SES</td>
<td>10.1.1.84</td>
</tr>
<tr>
<td>SES</td>
<td>10.1.1.84</td>
<td>val1-tn2501ap</td>
<td>. . .</td>
</tr>
<tr>
<td>val1-tn2501ap</td>
<td>10.1.1.84</td>
<td></td>
<td>. . .</td>
</tr>
</tbody>
</table>

(11 of 11 administered node-names were displayed)

Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name

**Figure 7: IP Nodes Names Form**

**Step 3: Define IP Network Region**

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

- **The Authoritative Domain** field is configured to match the domain name configured on the Avaya SES. In this configuration, the domain name is *devcon.com*. This field is required for endpoints to call the public network.
- By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between SIP endpoints without using media resources such as the TN2302AP IP Media Processor (MedPro) card.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this configuration, this codec set will apply to calls with PAETEC Communications as well as any IP phone (H.323 or SIP) within the enterprise.

In this case, the SIP trunk is assigned to the same IP network region as the G650 Media Gateway, CLAN and MedPro cards. If multiple network regions are used, Page 3 of each *IP Network Region* form must be used to specify the codec set for inter-region communications.
Note also that the **IP Network Region** form is used to set the packet parameters that provides priority treatment for signaling and audio packets over other data traffic on PAETEC Communications access facilities. These parameters may need to be aligned with the specific values provided by PAETEC Communications.

```change ip-network-region 1
Region: 1
Location: Authoritative Domain: devcon.com
Name:

MEDIA PARAMETERS
  Codec Set: 1
    Intra-region IP-IP Direct Audio: yes
    Inter-region IP-IP Direct Audio: yes
    IP Audio Hairpinning? y
    UDP Port Min: 2048
    UDP Port Max: 3028
    RTCP Reporting Enabled? y
    Call Control PHB Value: 46
    Audio PHB Value: 46
    Video PHB Value: 26
    RTCP MONITOR SERVER PARAMETERS
    Use Default Server Parameters? y
    Call Control 802.1p Priority: 6
    Audio 802.1p Priority: 6
    H.323 IP ENDPOINTS
    Idle Traffic Interval (sec): 20
    Keep-Alive Interval (sec): 5
    Keep-Alive Count: 5
```

**Figure 8: IP Network Region Form**

**Step 4: Define IP Codecs**

Open the **IP Codec Set** form using the **Codec Set** value specified in the **IP Network Region** form (Figure 8) and enter the audio codec type to be used for calls routed over the SIP trunk. Typical settings of the **IP Codec Set** form are shown in Figure 9. Note that the **IP Codec Set** form may include multiple codecs listed in priority order to allow the codec for the call to be negotiated during call establishment. For PAETEC Communications, only G.711MU, G.711A and/or G.729B can be included in this list.

```change ip-codec-set 1
Codec Set: 1

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

**Figure 9: IP Codec Set Form**
Step 5: Configure the Signaling Group

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

- Set the Group Type field to sip.
- The Transport Method field will default to tls (Transport Layer Security). TLS is the only link protocol that is supported for SIP trunking with Avaya SIP Enablement Services.
- Specify the Avaya CLAN card (node name “CLAN”) and the Avaya SIP Enablement Services Server (node name “SES”) as the two ends of the signaling group in the Near-end Node Name and the Far-end Node Name fields, respectively. These field values are taken from the IP Node Names form shown in Figure 7. For smaller media server platforms, the near (local) end of the SIP signaling group may be the S8300 media server processor (procr) rather than the CLAN.
- Ensure that the recommended TLS port value of 5061 is configured in the Near-end Listen Port and the Far-end Listen Port fields.
- Enter the IP Network Region value assigned in the ip-network-region form (Figure 8) into the Far-end Network Region field.
  Note that if the Far-end Network Region field is different from the near-end network region, the preferred codec will be selected from the IP codec set assigned for the inter-region connectivity for the pair of network regions. In this case, the same ip network region (Network Region 1) was used for local and PSTN calls; however, different network regions can be used in the field.
- Enter the domain name of Avaya SIP Enablement Services in the Far-end Domain field. In this configuration, the domain name is devcon.com. This domain is specified in the Uniform Resource Identifier (URI) of the SIP “To” address in the INVITE message. Mis-configuring this field may prevent calls from being successfully established to other SIP endpoints or to the PSTN.
- If calls to/from SIP endpoints are to be shuffled, then the Direct IP-IP Audio Connections field must be set to ‘y’.
- The DTMF over IP field should remain set to the default value of rtp-payload. This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833.
- The default values for the other fields may be used.
Step 6: Configure the Trunk Group

Configure the **Trunk Group** form as shown in **Figure 11** using the “add trunk-group” command. In this case the trunk group number chosen is 1. On Page 1 of this form:

- Set the **Group Type** field to *sip*.
- Choose a mnemonic **Group Name**.
- Specify an available trunk access code (TAC).
- Set the **Service Type** field to *tie*.
- Specify the signaling group associated with this trunk group in the **Signaling Group** field as previously specified in **Figure 10**.
- Specify the **Number of Members** supported by this SIP trunk group.

Note that one trunk member is required for each call between a non-SIP endpoint and PAETEC Communications. Calls involving a SIP endpoint and PAETEC Communications will use two trunk members for the duration of the call.
add trunk-group 1

TRUNK GROUP

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

TRUNK PARAMETERS

Unicode Name? y
Redirect On OPTIM Failure: 5000
SCCAN? n
Digital Loss Group: 18

Figure 11: Trunk Group Form – Page 1

On Page 2 of the Trunk Group form:
- Set the Numbering Format field to public. This field specifies the format of the calling party number sent to the far-end.

add trunk-group 1

TRUNK FEATURES

ACA Assignment? n
Measured: none
Maintenance Tests? y
Numbering Format: public
Replace Unavailable Numbers? n

Figure 12: Trunk Group Form – Page 2

Step 7: Configure Calling Party Number Information
Configure the Numbering Public/Unknown Format form to send the full calling party number to the far-end.

In this case, all stations with a 5-digit extension 7xxxx should send the calling party number 585-307-xxxx when an outbound call uses SIP trunk group #1. This calling party number will be sent to the far-end in the SIP “From” header.
Figure 13 shows the use of the “change public-unknown numbering” command to implement this rule.

<table>
<thead>
<tr>
<th>Ext</th>
<th>Len</th>
<th>Code</th>
<th>CPN</th>
<th>Total</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ext</td>
<td>Len</td>
<td>Code</td>
<td>CPN</td>
<td>Total</td>
</tr>
<tr>
<td>Trk</td>
<td>Ext</td>
<td>CPN</td>
<td>CPN Len</td>
<td>Ext Code</td>
</tr>
<tr>
<td>5</td>
<td>7</td>
<td>1</td>
<td>58530</td>
<td>10</td>
</tr>
</tbody>
</table>

**Figure 13: Numbering Public/Unknown Format Form**

### Step 8: Configure Incoming Digit Translation

This step performs the steps necessary to map incoming DID calls to the proper extension(s).

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

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

To create a fully mapped extension number as shown in **Figure 14**:

- Open the **Incoming Call Handling Treatment** form for the SIP trunk group.
- For each extension assigned a DID number from PAETEC Communications, enter 10 into the Called Len and Del fields, and the entire 10 digit DID number into the Called Number field. Enter the desired Avaya Communication Manager extension number into the Insert field.

<table>
<thead>
<tr>
<th>Service/Feature</th>
<th>Called Len</th>
<th>Called Number</th>
<th>Del</th>
<th>Insert</th>
</tr>
</thead>
<tbody>
<tr>
<td>tie</td>
<td>10</td>
<td>5853079960</td>
<td>10</td>
<td>70000</td>
</tr>
<tr>
<td>tie</td>
<td>10</td>
<td>5853079961</td>
<td>10</td>
<td>71001</td>
</tr>
<tr>
<td>tie</td>
<td>10</td>
<td>5853079962</td>
<td>10</td>
<td>71002</td>
</tr>
</tbody>
</table>

**Figure 14: Incoming Call Handling Treatment – Full Extension Mapping**

If the customer’s extension numbering plan aligns with the DID numbers (i.e., the final DID digits match the extension), it is not necessary to define an entry for each DID number.

Assuming a PBX dial plan that used the 5 digit extensions 71000 thru 71999 and assuming PAETEC Communications provided DID numbers of 585-307-1000 thru 1999, the incoming number translation would be done similar to **Figure 15**. Note that the Called Number entry in this case represents the common matching portion applicable to all incoming numbers. Thus 5853071 matches all numbers in the assigned DID block from PAETEC Communications.
INCOMING CALL HANDLING TREATMENT

<table>
<thead>
<tr>
<th>Feature</th>
<th>Len</th>
<th>Called Number</th>
<th>Del</th>
<th>Insert</th>
</tr>
</thead>
<tbody>
<tr>
<td>tie</td>
<td>10</td>
<td>5853071</td>
<td>5</td>
<td></td>
</tr>
</tbody>
</table>

**Figure 15: Incoming Call Handling Treatment – Simple Extension Mapping**

**Step 9: Save Avaya Communication Manager Changes**

Enter the “save translation” command to make the changes permanent.

### 3.2. SIP Endpoint Configuration

This section describes the administration of SIP telephones and requires the preceding SIP Trunk configuration to have been completed. SIP telephones are optional and not required to use the PAETEC Communications IPATH Service.

**Step 1: Assign a Station**

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

Using the “add station” command from the SAT:

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

The remaining fields are configured per normal station administration that is beyond the scope of these Application Notes. Note that the Class of Restrictions (**COR**) and Class of Service (**COS**) will govern the features and call restrictions that apply to this station.
On Page 2 of the Station form:

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

Setting the **Restrict Last Appearance** value to ‘y’ reserves the last call appearance for outbound calls. Certain SIP conference and transfer features will not function properly if a third appearance is not available for incoming calls.
On Page 3 of the Station form, configure at least 3 call appearances for the SIP telephone as shown in Figure 18.

![Station Administration – Page 3](image)

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

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

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

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

- Specify the Station Extension of the SIP endpoint.
- Set the Application field to OPS.
- Set the Phone Number field to the digits to be sent over the SIP trunk. In this case, the SIP telephone extensions configured on Avaya SIP Enablement Services also match the extensions of the corresponding AWOH stations on Avaya Communication Manager. However, this is not a requirement.
- Set the Trunk Selection field to ‘1’, which is the number assigned to the SIP trunk group used to route the call to the SIP station. This trunk group number was previous defined in Figure 11.
- Set the Configuration Set value. In these Application Notes, Configuration Set 1 uses the default values of the Configuration Set form.
On Page 2, set the Call Limit field to the maximum number of calls that may be active simultaneously at the station. In this example, the call limit is set to ‘3’, which corresponds to the number of call appearances configured on the station form. Accept the default values for the other fields.

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

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

4. Configure Avaya SIP Enablement Services
This section covers the administration of Avaya SIP Enablement Services. Avaya SIP Enablement Services is configured via an Internet browser using the Administration web interface. It is assumed that Avaya SIP Enablement Services software and the license file have already been installed. During the software installation, the install script is run on the Linux shell of the server to specify the IP network properties of the server along with other parameters. For additional information on these installation tasks, refer to [4].

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

4.1. SIP Trunking to PAETEC Communications
Step 1: Log in to Avaya SIP Enablement Services
Access the SES Administration web interface, by entering http://<ip-addr>/admin as the URL in an Internet browser, where <ip-addr> is the IP address of Avaya SIP Enablement Services server.
Log in with the appropriate credentials and then select the **Launch Administration Web Interface** link from the main screen as shown in **Figure 21**.

**Figure 21 - Avaya SES Main Screen**

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

**Figure 22: Avaya SES Administration Home Page**
Step 2: Define System Properties
From the left pane of the Administration web interface, expand the Server Configuration option and select System Properties. This screen displays the SES version and the network properties entered via the install script during the installation process.

In the Edit System Properties screen,

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

![System Properties](image)

**Figure 23: System Properties**
Step 3: Enter Avaya SES Host Information
After setting up the domain in the Edit System Properties screen, create a host computer entry for Avaya SIP Enablement Services. The following example shows the Edit Host screen since the host had already been added to the system.

The Edit Host screen shown in Figure 24 is accessible by clicking on the Hosts link in the left pane and then clicking on the Edit option under the Commands section of the subsequent page that is displayed.

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

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

In the Add Media Server screen, enter the following information:
- A descriptive name in the Media Server Interface field (e.g., S8710-CLAN).
- Select the home SES server in the Host field as specified in Figure 24.
- Select TLS (Transport Link Security) for the Link Type. TLS provides encryption at the transport layer. TLS is the only link protocol that is supported for SIP trunking with Avaya Communication Manager.
- Enter the IP address of the Avaya S8710 Media Server CLAN board in the **SIP Trunk IP Address** field. (Note: This may be the IP address of the media server processor in smaller Avaya Communication Manager configurations using an Avaya S8300 Media Server.)
- After completing the **Add Media Server** screen, click on the **Add** button.

![Add Media Server](image)

**Figure 25: Add Media Server**

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

This routing compares the Uniform Resource Identifier (URI) of an incoming INVITE message to the pattern configured in the Media Server Address Map, and if there is a match, the call is routed to the designated Avaya Communication Manager. The URI usually takes the form of `sip:user@domain`, where *domain* can be a domain name or an IP address. Patterns must be specific enough to uniquely route incoming calls to the proper destination if there are multiple Avaya Communication Manager systems supported by the Avaya SES server.
In these Application Notes, only incoming calls from the PSTN require a media server address map entry. Calls originated by Avaya SIP telephones configured as OPS are automatically routed to the proper Avaya Communication Manager by the assignment of an Avaya Media Server extension to that phone. Address map definitions for SIP endpoints not assigned a media server extension and connections to multiple service providers are beyond the scope of these Application Notes.

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

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

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

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

The strategy used to define the media server address maps will be to create a pattern that matches the DID numbers assigned to the customer by PAETEC Communications. The SES will forward the messages with matching patterns to the appropriate CLAN interface of the S8710 media server.

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

- Select **Media Servers** in the left pane of the Administration web interface. This will display the **List Media Servers** screen.
- Click on the **Map** link associated with the appropriate media server to display the **List Media Server Address Map** screen.
- Click on the **Add Map In New Group** link. The screen shown in **Figure 26** is displayed. The **Host** field displays the name of the media server that this map applies to.
- Enter a descriptive name in the **Name** field.
- Enter the regular expression to be used for the pattern matching in the **Pattern** field.

In these Application Notes, the DID numbers provided by PAETEC Communications are 585-307-9900 thru 9999. The pattern specification (without the double quotes) for DID numbers assigned is: `^sip:58530799[0-9]{2}`. This means that URIs beginning with “sip:58530799” followed by any other 2 digits will match the pattern and be routed to the interface defined for S8710-CLAN.

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

- Click the **Add** button once the form is completed.
After configuring the media server address map, the **List Media Server Address Map** screen appears as shown in **Figure 27**.
Note that after the first **Media Server Address Map** is added, the **Media Server Contact** is created automatically. For the **Media Server Address Map** added in Figure 26, the following contact was created:

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

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

**Step 6: Specify Address Maps to PAETEC Communications**

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

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

In this configuration, the Avaya SES routing rule for the SIP trunk group will be to send all outbound PSTN traffic to the PAETEC Communications IPATH Service. To perform this, several dialing patterns will be created in the Avaya SES.

- The first pattern (without the double quotes) of “^sip:1[0-9]{10}” will match on all sip calls having 1 followed by any 10 digits.
- The second pattern of “^sip:0” will route any sip call beginning with 0 (regardless of the following digits).
- Finally N11 service codes (such as 411, 611, etc.) will be recognized using the pattern “^sip:[2-9]11”.

Note that additional or more specific pattern matches would be used if necessary to selectively route SIP traffic to different destinations (such as multiple service providers serving different geographic regions). Also note that a user dialed access code (such as 9 to place a PSTN call) has been previously deleted (by ARS) prior to seizing the outbound SIP trunk.
The configuration of the host address map for all 1 plus 10-digit North American calls is shown in Figure 28.

- Access the Add Host Address Map screen by selecting the Hosts link in the left pane of the Administration web interface and then clicking on the Map link associated with the appropriate host (e.g., k2.devcon.com). The List Host Address Map screen is displayed.
- From this screen, click the Add Map In New Group link to display the Add Host Address Map screen shown in Figure 28.
- Enter a descriptive name for the map, such as “PAETEC_1Plus10”.
- Specify an appropriate pattern for the call type. In this example, the pattern used for North American calls is “^sip:1[0-9]{10}”.
- Leave the Replace URI checkbox selected.
- Click the Add button.

![Figure 28: Edit Host Map Entry](image)

Figure 28: Edit Host Map Entry
Additional Host Address Map patterns are added in a similar manner. Figure 29 illustrates the entry for Operator “zero” and “zero-plus” dialing.

![Figure 29: Address Map for 0 and 0+ Dialing](image)
Figure 30 illustrates the host address map for the N11 service codes.

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

To enter the PAETEC Communications SIP gateway information:
- As described in Step 6, display the List Host Address Map screen.
- Click on the Add Another Contact link associated with the address map added in Figure 28 and Figure 29 to open the Add Host Contact screen. In this screen, the Contact field specifies the destination for the call and it is entered as:

  \[ \text{sip:$(user)@20.1.1.135:5060;transport=udp} \]

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

After configuring the host address maps and contact information, the List Host Address Map screen will appear as shown in Figure 31.
Step 8: Save the Changes
After making changes within Avaya SES, it is necessary to commit the database changes using the Update link that appears when changes are pending. Perform this step by clicking on the Update link found in the bottom of the blue navigation bar on the left side of any of the SES Administration screens as shown in Figure 32.
Step 9: Specify the PAETEC Communications SIP Gateway as a Trusted Host
The final step to complete the SIP trunk administration on Avaya SES is to designate the IP address of PAETEC Communications SIP Gateway as a trusted host. As a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the designated IP address.¹

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

To configure a trusted host:
- Log in to the Avaya SES using the administrative login and password.
- Enter the following trustedhost command at the Linux shell prompt:
  
  \[ \text{trustedhost} \ -a \ 20.1.1.135 \ -n \ k2.devcon.com \ -c \ PAETEC\_Gway \]
  
  The \(-a\) argument specifies the address to be trusted; \(-n\) specifies the SES host name; \(-c\) adds a comment.
- Use the following trustedhost command to verify the entry is correct:
  
  \[ \text{trustedhost} \ -L \]
  
  Figure 33 illustrates the results of the trustedhost commands.²
- Complete the trusted host configuration by returning to the main Avaya SES Administration web page and again clicking on the **Update** link as shown in Figure 32.

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

---

¹ Note, if the trusted host step is not done, authentication challenges to incoming SIP messages (such as INVITEs and BYEs) will be issued by the SES. This may cause call setup to fail, active calls to be disconnected after timeout periods, and/or SIP protocol errors.

² For completeness, the \(-d\) argument allows the trust relationship to be deleted. For example,

  \[ \text{trustedhost} \ -d \ 20.1.1.135 \ -n \ k2.devcon.com \]

removes the trust relationship added above.
admin@k2> trustedhost -a 20.1.1.135 -n k2.devcon.com -c PAETEC_Gway
20.1.1.135 is added to trusted host list.

admin@k2> trustedhost -L
Third party trusted hosts.
<table>
<thead>
<tr>
<th>Trusted Host</th>
<th>CCS Host Name</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>20.1.1.135</td>
<td>k2.devcon.com</td>
<td>PAETEC_Gway</td>
</tr>
</tbody>
</table>

Figure 33: Configuring a Trusted Host

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

4.2. Configuration for SIP Telephones

This section provides basic instructions for completing the administration necessary to support the optional Avaya 4600 Series SIP telephones. Additional features, such as the use of mnemonic addressing, are beyond the scope of these Application Notes.

Step 1: Add a SIP User

Create the SIP user record as follows:

- In Avaya SES administration, expand the Users link in the left side blue navigation bar and click on the Add link.
- In the Add User screen, enter the extension of the SIP endpoint in the Primary Handle field.
- Enter a user password in the Password and Confirm Password fields. This password will be used when logging into the user’s SIP telephone.
- In the Host field, select the Avaya SIP Enablement Services server hosting the domain (devcon.com) for this user. Enter the First Name and Last Name of the user.
- To associate a media server extension with this user, select the Add Media Server Extension checkbox. Calls from this user will always be routed through Avaya Communication Manager over the SIP trunk for origination services.
- Press the Add button. This will cause a confirmation screen to appear.
- Press Continue on the confirmation screen.
Step 2: Specify Corresponding Avaya Communication Manager Extension

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

- In the Add Media Server Extension screen, enter the Extension configured on the media server, shown in Figure 16, for the OPS extension on Avaya Communication Manager previously defined in Section 3.2. Usually, the media server extension and the user extension are the same (recommended) but it is not required to be.
- Select the Media Server assigned to this extension.
- Click on the Add button.
- To commit the configuration changes, click on the Update link in the left pane.
Step 3: Repeat for Each SIP User
Repeat Steps 1 and 2 for each SIP user.

5. PAETEC Communications IPATH Service Configuration
In order to use PAETEC Communications IPATH Service, a customer must request service from PAETEC using their sales processes. The process can be started by contacting PAETEC Communications via their corporate web site at http://www.paetec.com/contact/inforequest.asp and requesting information via the online sales links or telephone numbers.
During the signup process, PAETEC will require that the customer provide the public IP address used to reach the Avaya SIP Enablement Services server. (Note the address used within these Application Notes is 10.1.1.86; the actual IP address will be specific to the customer implementation).

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

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

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

6. Interoperability Compliance Testing

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

6.1. General Test Approach

A simulated enterprise site consisting of an Avaya SIP Telephony solution supporting SIP trunking was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the commercially available IPATH SIP Trunking Service provided by PAETEC Communications. This allowed the enterprise site to use SIP trunking for PSTN calling.

The following features and functionality were covered during the SIP trunking interoperability compliance test:

- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by PAETEC Communications.
- Outgoing calls from the enterprise site were completed via PAETEC Communications to the PSTN destinations.
- Calls using SIP, H.323, digital and analog endpoints supported by the Avaya SIP Telephony solution.
- Various call types including: local, long distance, international, toll free, operator and directory assistance calls.
- Calls using G.711mu and G.729B codecs.
- DTMF transmission using RFC 2833.
- Telephone features such as hold, transfer and conference.
- Voicemail coverage and retrieval for endpoints at the enterprise site.
- Direct IP-to-IP media (also known as “shuffling”) which allows SIP endpoints to send audio (RTP) packets directly to each other without using media resources on the Avaya Media Gateway.
6.2. Test Results
All tests were completed successfully.

7. Verification Steps
This section provides the verification steps that may be performed in the field to verify that the SIP, H.323, digital and analog endpoints can place outbound and receive inbound PSTN calls through PAETEC Communications.

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

8. Support
For technical support on PAETEC Communications IPATH Service, contact PAETEC Communications Customer Service at 877-340-2600 or customerservice@PAETEC.com Include the customer account number in the communication.

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

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


Additional information about PAETECH Communications IPATH Service is available at http://www.paetec.com.
APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by PAETEC Communications and the Avaya SIP network at the enterprise site. Customers may use these INVITE messages for comparison and troubleshooting purposes. Differences in these messages may indicate different configuration options selected.

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

INVITE sip:5853079961;npdi@10.1.1.86;user=phone SIP/2.0
Via: SIP/2.0/UDP 20.1.1.135:5060;branch=z9hG4bK0odoaj0070ag0akrt7o8.1
From: "AVAYA INC C/O T"<sip:7328571637@20.1.1.135:5060;user=phone;avaya1plex=AVAYA1PLEX-q2jm5jv0cfg77;tag=10000000-0-1108989597
To: <sip:5853079961@10.1.1.86;user=phone>
CSeq: 1 INVITE
Contact: "AVAYA INC C/O T"<sip:7328571637@10.1.1.86;avaya1plex=AVAYA1PLEX-q2jm5jv0cfg77;transport=udp>
Call-ID: 10DA5B20-4E7B6@10.254.1.7
P-Asserted-Identity: "AVAYA INC C/O T"<sip:7328571637@10.254.1.7;user=phone>
Privacy: none
Max-Forwards: 69
Content-Type: application/sdp
Content-Length: 218

v=0
o=- 3355764663 3355764663 IN IP4 20.1.1.135
s=-
c=IN IP4 20.1.1.135
t=0 0
m=audio 20038 RTP/AVP 0 8 18 13
a=ptime:20
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:13 CN/8000
Sample SIP INVITE Message from Avaya SIP Enablement Services to PAETEC Communications:

```
INVITE sip:15853402870@20.1.1.135:5060;transport=udp SIP/2.0
Call-ID: 0e21859e20db1302444ce3fad00
CSeq: 1 INVITE
From: "Dcp Phone62004" <sip:15853079963@east.devcon.com:5061>;tag=0e21859e20db12f2444ce3fad00
Record-Route: <sip:10.1.1.86:5060;lr>,<sip:10.1.1.84:5061;lr;transport=tls>
To: "15853402870" <sip:15853402870@east.devcon.com>
Via: SIP/2.0/UDP 10.1.1.86:5060;branch=z9hG4bK03030356656232323333f.0,SIP/2.0/TLS
10.1.1.84;psrrposn=2;branch=z9hG4bK0e21859e20db1312444ce3fad00
Content-Length: 201
Content-Type: application/sdp
Contact: "Dcp Phone62004" <sip:15853079963@10.1.1.84:5061;transport=tls>
Max-Forwards: 69
User-Agent: Avaya CM/R013x.01.0.628.6
Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
Session-Expires: 1800;refresher=uac
Min-SE: 1800
History-Info: <sip:15853079963@east.devcon.com>;index=1
History-Info: "15853402870" <sip:15853402870@east.devcon.com>;index=1.1
Supported: 100rel,timer,replaces,join,histinfo
P-Asserted-Identity:"Dcp Phone62004"<sip:15853079963@east.devcon.com:5061>

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

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

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

In the pattern matching string used in the Avaya SES:

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

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

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

This reads as: “Strings that begin with exactly `sip:1` and having any 10 digits following will match.

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

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