

Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between BandTel Communications Global SIP Trunking Service with an Avaya IP Telephony Solution – 1.0

Abstract

These Application Notes describe the steps to configure SIP trunking between BandTel Global SIP Trunking service and an Avaya IP Telephony solution. The Avaya solution consists of Avaya SIP Enablement Services, Avaya Communication Manager, and various Avaya SIP, H.323, digital and analog endpoints.

BandTel is a California based Global VoIP service provider. They are committed to offering VoIP telephony solutions to high volume telecom users worldwide such as Call Centers, Enterprise users, Teleconferencing companies and Internet Voice Response (IVR) users. Their Global SIP Trunking service allows customers with an Avaya SIP-based solution to place and receive calls to and from the PSTN. This includes outbound local, long distance, and international calling, as well as inbound calling to Direct Inward Dialing (DID) numbers. SIP trunking 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.

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

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1. Introduction

These Application Notes describe the steps for configuring SIP (Session Initiation Protocol) trunking between the BandTel Global SIP Trunking service and an Avaya IP telephony solution consisting of Avaya SIP Enablement Services, Avaya Communication Manager and various Avaya telephony endpoints. These endpoints included IP telephones (using SIP and H.323 protocols), traditional analog and digital phones and a SIP soft phone known as the Avaya One-X Desktop Edition running on a Microsoft Windows PC. BandTel is a California based Global VoIP service provider. They are committed to offering VoIP telephony solutions to high volume telecom users worldwide such as Call Centers, Enterprise users, Teleconferencing companies and Internet Voice Response (IVR) users.

BandTel's Global SIP Trunking network offers a highly robust and redundant network architecture called the "N Plus Matrix". There are N Plus SIP soft switches deployed and active at all times sharing load as well as being ready to handle load from another node in the event of a catastrophic failure.

Customers using BandTel's Global SIP Trunking service are able to place and receive PSTN calls via a dedicated broadband Internet connection using SIP. This converged network solution is a flexible, cost-saving alternative to more traditional PSTN trunks such as T1 or ISDN PRI.

BandTel's Enhanced Service offers the following capabilities:
• Outbound Domestic calling to local and long distance services
• Outbound International calling
• Incoming Direct Inward Dial (DID) service
• 800 Domestic service

Note: BandTel appends a unique four digit prefix to the beginning of the dialed number in order to identify a standard DID number or an 8xx number. The prefix "0300" is used for standard DID numbers while the prefix "0200" is used for 8xx numbers.

Figure 1 illustrates an example Avaya IP telephony solution connected to BandTel's Global SIP Trunking service. This is the configuration used during the Developer*Connection* compliance testing process.

- The Avaya IP telephony solution used to create a simulated customer site contained: 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 4620 IP telephone (configured to use either the SIP or H.323 protocol).
- Avaya 6416 digital and 6210 analog telephones.

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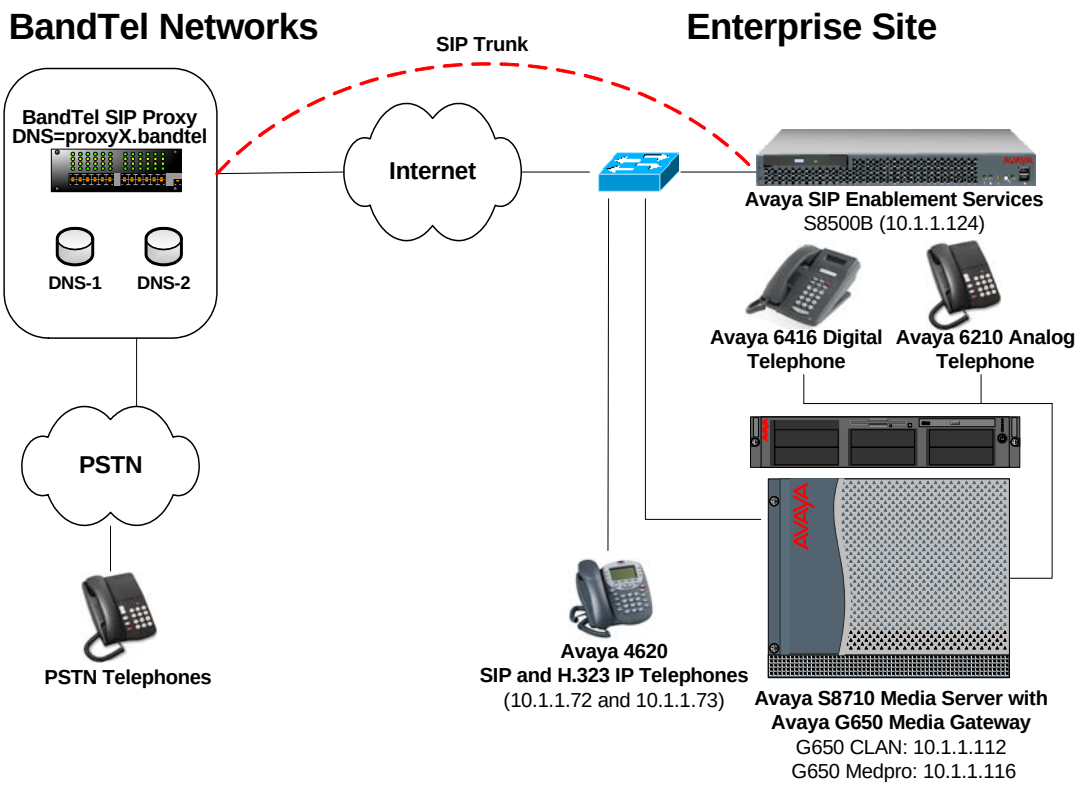


Figure 1: Avaya IP Telephony Network using BandTel Global SIP Trunking Service

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1.1 Call Flows

To better understand how calls are routed between the PSTN and the enterprise site shown in **Figure 1** using SIP trunks, two call flows are described in this section. The first call scenario illustrated in **Figure 2** is a PSTN call to the enterprise site terminating on a typical analog telephone supported by Avaya Communication Manager.

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

2. Based on the DID number, BandTel 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 between the elements.

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

- or –

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

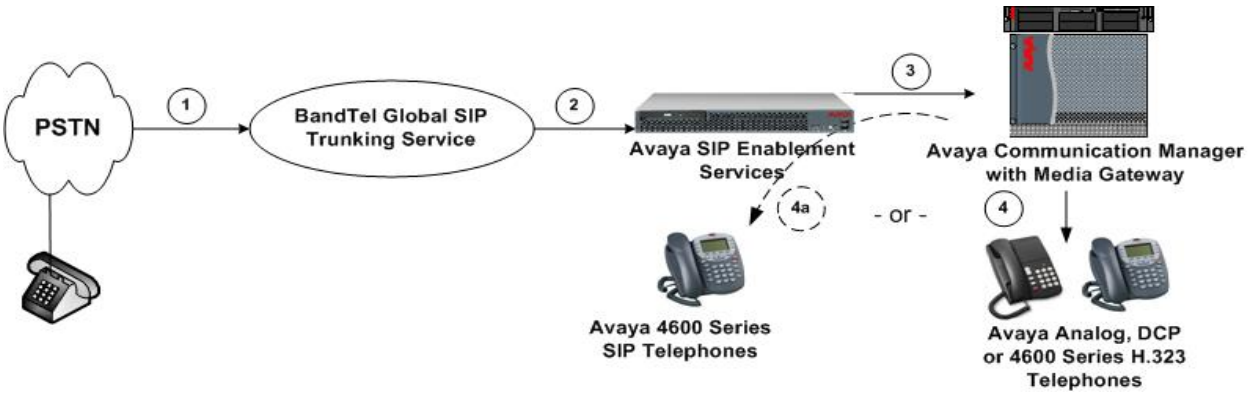


Figure 2: Incoming PSTN Calls to Avaya Communication Manager

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

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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 BandTel.

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 BandTel.

4. BandTel completes the call to the PSTN.

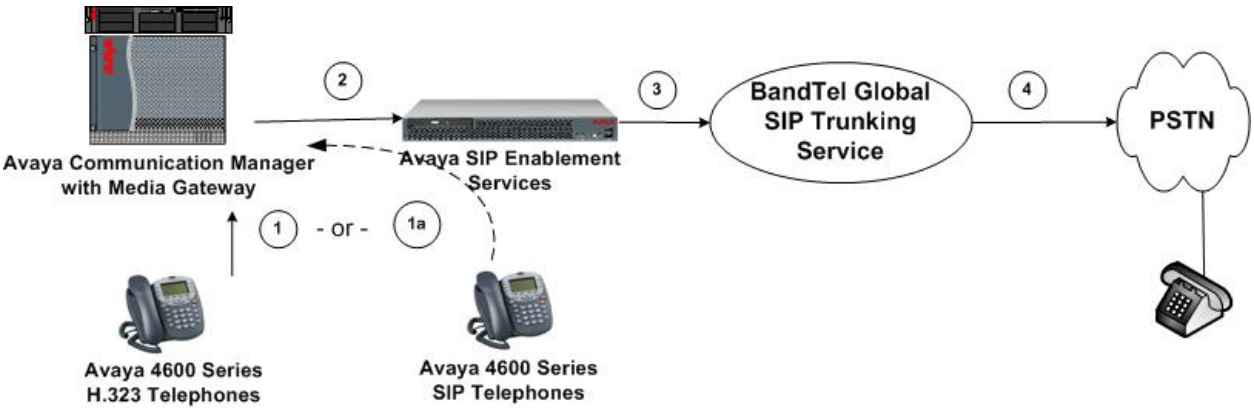


Figure 3: Outgoing Calls from Avaya Communication Manager to the PSTN

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2. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Avaya S8710 Media Server with an Avaya G650 Media Gateway	Communication Manager 3.1.2 R013x.01.2.632.1 Update: 01.2.632.1-12866
Avaya SIP Enablement Services	SES-3.1.0.0-018.0 SES 3.1 Service Pack 10013
Avaya 4620 IP Telephone	R2.2.2 – SIP (s10d0b2.2.2.bin)
Avaya one-X Desktop Edition SIP endpoint	R2.1 – SIP
Avaya 4620 IP Telephone	R2.3 – H.323 (a10d01b2_3.bin)
Avaya 6416 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
BandTel Global SIP Trunking Service Components	
BandTel SIP Trunking Service Network	Version 12.6

Table 1: Equipment and Software Tested

The specific configuration above was used for the BandTel 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 (SES) server. This trunk will carry the SIP signaling sent to the BandTel Global SIP Trunking service.

This SIP trunk also provides the trunking for SIP endpoint devices such as Avaya 4620 SIP Telephone and Avaya SIP Softphone using Avaya Communication Manager in the recommended off-PBX stations (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.

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Note the use of SIP endpoints is optional. The steps discussed in Sections 3.2 and 4.3 describing SIP endpoint administration may be omitted if SIP endpoints are not used. In the Avaya SIP architecture, the Avaya SES acts as a SIP proxy through which all incoming and outgoing SIP messages flow to BandTel. There is no direct SIP signaling path between BandTel 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 BandTel 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, 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.

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.

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3.1 Sip Trunk Configuration

Step 1: Confirm Necessary Optional Features

Log into the Avaya Communication Manager's SAT interface and confirm that sufficient SIP trunk and Off PBX Telephone capacities are enabled. 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: V13
      Location: 1                              RFA System ID (SID): 1
      Platform: 8                              RFA Module ID (MID): 1

                                                          USED
                                  Platform Maximum Ports: 44000 229
                                       Maximum Stations: 36000 53
                                Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 10    0
                    Maximum Off-PBX Telephones -   OPS: 36000 23
                    Maximum Off-PBX Telephones - SCCAN: 0     0


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

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

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On Page 2, verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient for the combination of trunks to the BandTel network, SIP endpoints and any other SIP trunks used. Note that each SIP OPS telephone on a call with BandTel uses two SIP trunks for the duration of the call.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 100     20
          Maximum Concurrently Registered IP Stations: 100     1
            Maximum Administered Remote Office Trunks: 0       0
Maximum Concurrently Registered Remote Office Stations: 0      0
             Maximum Concurrently Registered IP eCons: 0       0
  Max Concur Registered Unauthenticated H.323 Stations: 5      0
                 Maximum Video Capable H.323 Stations: 10      0
                  Maximum Video Capable IP Softphones: 10      0
                      Maximum Administered SIP Trunks: 200     156

   Maximum Number of DS1 Boards with Echo Cancellation: 1      0
                            Maximum TN2501 VAL Boards: 1      1
                     Maximum G250/G350/G700 VAL Sources: 50   0
          Maximum TN2602 Boards with 80 VoIP Channels: 2      1
         Maximum TN2602 Boards with 320 VoIP Channels: 2      0
   Maximum Number of Expanded Meet-me Conference Ports: 0      0

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

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

Step 2: Assign Node Names

In the **IP Node Names** form, assign the node name and IP address for the Avaya SIP Enablement Server at the enterprise site as shown in **Figure 6**. In this case "SES" and "10.1.1.124" 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 "10.1.1.112" are the name and IP address assigned to the TN799DP Control-Lan card. The CLAN entry was previously created during the installation of the system. Note, in smaller gateways such as an Avaya G350, the S8300 processor address (procr) is used as the SIP signaling interface instead of the CLAN interface.

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```
change node-names ip                                          Page   1 of   1
                                  IP NODE NAMES
    Name                 IP Address          Name              IP Address
CLAN                   10 .1  .1  .112                          .   .   .
default                0  .0  .0  .0                           .   .   .
ipsi                   10 .1  .1  .109                          .   .   .
medpro-hw11            10 .1  .1  .116                          .   .   .
procr                  10 .1  .1  .104                          .   .   .
SES                    10 .1  .1  .124                          .   .   .
val1-tn2501ap          10 .1  .1  .122                          .   .   .
windowPC               10 .1  .1  .101                          .   .   .
                          .   .   .                             .   .   .
                          .   .   .                             .   .   .
( 8 of 8   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

Figure 6: IP Nodes Names Form

Step 3: Define IP Network Region

The **IP Network Region** form specifies the parameters used by the SIP trunk group serving the Avaya SES server (used to reach BandTel and any optional SIP endpoints). Note that these parameters also apply to any other elements (such as H.323 phones, MedPro cards, etc.) also assigned to this region. Use the **change ip-network-region** command to set the following values:

- The **Authoritative Domain** field is configured to match the domain name configured on the Avaya SES. In this configuration, the domain name is *east.devcon.com*. This field is required for endpoints to call the public network.
- 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 media resources such as the TN2302AP IP Media Processor (MedPro) card. In the case of BandTel, **IP-IP Direct Audio** will not be supported and these parameters will be set to no. See section **6.2 Test Results** for a detailed explanation.
- 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 BandTel as well as any IP telephone (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 QoS packet parameters that provides priority treatment for signaling and audio packets over other data traffic on BandTel's Global SIP Trunking service. These parameters may need to be aligned with the specific values provided by BandTel.

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


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```
                              IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: east.devcon.com
    Name: Default System All
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: no
      Codec Set: 1              Inter-region IP-IP Direct Audio: no
   UDP Port Min: 2048                     IP Audio Hairpinning? y
   UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS                RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? n
        Video PHB Value: 26                  Server IP Address: 10 .1  .1  .112
802.1P/Q PARAMETERS                                Server Port: 5005
 Call Control 802.1p Priority: 6       RTCP Report Period(secs): 5
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                    RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Figure 7: IP Network Region Form

Step 4: Define IP Codecs

Open the **IP Codec Set** form using the ip-codec value specified in the **IP Network Region** form (**Figure 7**) and enter the audio codec type to be used for calls routed over the SIP trunk. The settings of the **IP Codec Set** form are shown in **Figure 8.** Note that the **IP Codec Set** form may include multiple codecs listed in priority order to allow the codec for the call to be negotiated during call establishment. Although G.729b can be offered by special request, BandTel recommends the G.711MU codec.

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

                         IP Codec Set

    Codec Set: 1

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

Figure 8: IP Codec Set Form

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Step 5: Configure the Signaling Groups

For interoperability with BandTel, two signaling groups must be configured. One signaling group will be used for outbound calls while the second signaling group will be used for inbound calls. This is necessary because BandTel requires that subscribers use a DNS name to reach BandTel's proxy server for outbound calls from an enterprise site into their network rather than an IP address. This requires the "Far End Domain" field on the signaling group form to be set to the BandTel proxy server's DNS name. While this allows outbound Avaya calls through the BandTel network to the PSTN, incoming calls will not be able to use this same signaling group because BandTel does not use this DNS name when it issues SIP Invite messages. Instead, the BandTel service uses an IP address. Since Avaya Communication Manager uses the caller's domain/IP address from the SIP Invite message to match with the "Far End Domain" of a signaling group, there would not be a match. When this happens, Avaya Communication Manager will look for a signaling group with a blank "Far End Domain" field and use this group. If this does not exist, the call will be routed to a random signaling group provided that others exist. In order for inbound calls to then be routed in a deterministic way, another signaling group must be configured with a blank "Far End Domain" field set. This second signaling group can be thought of as a default signaling group. The configuration steps below show how to configure both of these signaling groups.

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

- Set the **Group Type** field to *sip*.
- The **Transport Method** field will default to *tls* (Transport Layer Security). TLS is the only link protocol that is supported for SIP trunking with Avaya SIP Enablement Services.
- **Near-end Node Name** Specify the Avaya Control-Lan card (node name "CLAN"). This field value is taken from the **IP Node Names** form shown in **Figure 6**. 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.
- **Far-end Node Name** Specify the Avaya SIP Enablement Services (node name "SES"). This field value is taken from the **IP Node Names** form 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.
- Enter the IP Network Region value assigned in the IP Network Region form (**Figure 7**). Note that if the **Far-end Network Region** field is different from the near-end network region, the preferred codec will be selected from the IP codec set assigned for the interregional 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 the Avaya SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is *proxy2.bandtel*. 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.

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- The **Direct IP-IP Audio Connections** field must be set to 'n'. BandTel does not support the Avaya **Direct IP-IP Audio** feature. See section **6.2 Testing Results** for a detailed explanation.
- 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. [8]
- The default values for the other fields may be used.

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

                               SIGNALING GROUP

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


   Near-end Node Name: CLAN                 Far-end Node Name: SES
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1
       Far-end Domain: proxy2.bandtel

                                            Bypass If IP Threshold Exceeded? n

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

Figure 9: Outbound Signaling Group Form

Now, configure the *inbound* **Signaling Group** form following the same steps used for the outbound signaling group above with one exception, leave the **Far-end Domain** field blank as shown in **Figure 10**:

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

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



   Near-end Node Name: CLAN                 Far-end Node Name: SES
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1
       Far-end Domain:

                                            Bypass If IP Threshold Exceeded? n

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

Figure 10: Inbound Signaling Group Form

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Step 6: Configure the Trunk Groups
As described above in step 5, two trunks must also be configured. One trunk will be paired with the outbound signaling group and the other with the inbound signaling group.

Configure the *outbound* **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 *outbound* signaling group associated with this trunk group in the **Signaling Group** field as previously specified in **Figure 9**.
- Specify the **Number of Members** supported by this SIP trunk group.

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

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

Group Number: 1                  Group Type: sip           CDR Reports: y
  Group Name: BandTel SIP Out          COR: 1        TN: 1        TAC: 101
   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: 10
```

Figure 11: Trunk Group Form (Outbound) – Page 1

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On Page 3 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.

```
change trunk-group 1                                          Page   3 of  21

TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y


                      Numbering Format: public
                                              Prepend '+' to Calling Number? n


                                                Replace Unavailable Numbers? n
```

Figure 12: Trunk Group Form (Outbound) – Page 3

Now, configure the *inbound* **Trunk Group** form as shown in **Figure 13** using the **add trunk-group** command. In this case the trunk group number chosen is 2. On Page 1 of this form:

- Set the **Group Type** field to *sip*.
- Choose a mnemonic **Group Name**.
- Specify an available trunk access code (**TAC**).
- Set the **Service Type** field to *tie*.
- Specify the *inbound* 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 BandTel. Calls involving a SIP endpoint and BandTel will use two trunk members for the duration of the call.

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

Group Number: 1                    Group Type: sip            CDR Reports: y
  Group Name: BandTel SIP In            COR: 1        TN: 1        TAC: 102
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                          Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n

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

Figure 13: Trunk Group Form (Inbound) – Page 1

On Page 3 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.

```
change trunk-group 2                                          Page   3 of  21
```


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```
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                      Maintenance Tests? y


                     Numbering Format: public
                                            Prepend '+' to Calling Number? n

                                              Replace Unavailable Numbers? n
```

Figure 14: Trunk Group Form (Inbound) – Page 3

Step 7: Configure Calling Party Number Information
Use the **change public-unknown-numbering** command to configure Avaya Communication Manager to send the full calling party number to the far-end.

In this case, all stations with a 5-digit extension beginning with 6 should send the calling party number 732-85x-xxxx when an outbound call uses SIP trunk group #1 (*this is the outbound trunk group specified in* **Step 6**). This calling party number will be sent to the far-end in the SIP "From" header.

Figure 15 shows the use of the **change public-unknown-numbering** command to implement this rule.

```
change public-unknown-numbering 0                              Page   1 of   2

                       NUMBERING - PUBLIC/UNKNOWN FORMAT
                                    Total                                 Total
Ext Ext    Trk        CPN            CPN Ext Ext    Trk        CPN          CPN
Len Code   Grp(s)     Prefix         Len Len Code   Grp(s)     Prefix       Len

 5  6       1          73285          10
```

Figure 15: Numbering Public/Unknown Format Form

Step 8: Automatic Route Selection for Outbound Calls
In these Application Notes, the Automatic Route Selection (ARS) feature will be used to route outbound calls via the SIP trunk to the BandTel Global SIP Trunking service to a PTSN destination.

Use the **change dialplan analysis** command to add ***9*** as a feature access code (**fac**).

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```
change dialplan analysis                                    Page   1 of  12
                           DIAL PLAN ANALYSIS TABLE
                                                          Percent Full:    3

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

Figure 16: Change Dialplan Analysis Form

Use the **change feature-access-codes** command to specify ***9*** as the access code for outside dialing.

```
change feature-access-codes                                 Page   1 of   7
                            FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: *03
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code:
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                  Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA: *10    All: *11    Deactivation: #10
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                   CDR Account Code Access Code:
                         Change COR Access Code:
                    Change Coverage Access Code:
                    Contact Closure   Open Code:          Close Code:
                    Contact Closure  Pulse Code:
```

Figure 17: Feature Access Codes Form

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Use the **change ars analysis** command to configure the route pattern selection rule based upon the number dialed following the dialed digit "9". In this sample configuration, the PSTN numbers dialed are all in the form 1AAANNXXXXX (A= Area Code, N=[2-9], X=[0-9]). If the area code (AAA) is 732, the call is to be routed to a route pattern containing the SIP trunk groups used for BandTel. Note that further administration of ARS is beyond the scope of these Application Notes but discussed in References [1] and [2].

```
change ars analysis 173                                    Page   1 of   2
                           ARS DIGIT ANALYSIS TABLE
                                  Location:  all           Percent Full:    3

         Dialed           Total       Route    Call   Node  ANI
         String            Min  Max  Pattern   Type   Num   Reqd
   173                     11   11     1       fnpa           n
```

Figure 18: ARS Analysis Form

Use the **change route-pattern** command to define the SIP trunk group included in the route pattern that ARS selects. In this configuration, route pattern 1 will be used to route calls to trunk group 1 (the SIP trunk created in **Step 6, Figure 11**).

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: To PSTN
                                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: 1    0       1                                                   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 3 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 19: Route Pattern Form

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Step 9: Configure Incoming Digit Translation
This step configures the settings necessary to map incoming DID calls to the proper extension(s).

The incoming digits sent in the INVITE message from BandTel are manipulated as necessary to route calls to the proper extension on Avaya Communication Manager. Note that this step assumes that the BandTel subscriber has already received their DID numbers from BandTel.

In the example used in these Application Notes, the incoming DID numbers provided by BandTel 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 20**:

- Open the **Incoming Call Handling Treatment** form for the *inbound* SIP trunk group configured in **Figure 13**, in this case Trunk Group 2.
- For each extension assigned a DID number from BandTel, enter **14** into the **Called Len** and **Del** fields, and the entire **14 digit DID number** into the **Called Number** field. (*BandTel appends a 4 digit prefix, (0300 or 0200), for identifying US DIDs and 8xx numbers*)
  Enter the desired Avaya Communication Manager extension number into the **Insert** field.

```
change inc-call-handling-trmt trunk-group 2                    Page   1 of  30

                        INCOMING CALL HANDLING TREATMENT
 Service/       Called    Called        Del  Insert
 Feature         Len      Number
 tie              14  02008662170590    14   61001
 tie              14  03007326269601    14   61001
 tie              14  03007326269602    14   60000
 tie              14  03007326269603    14   64003
 tie              14  03007326269604    14   60003
```

Figure 20: Incoming Call Handling Treatment – Full Extension Mapping

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If the customer's extension numbering 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 60000 thru 61999 and assuming BandTel provided DID numbers of 732-626-0000 thru 9999, the incoming number translation would be done similar to **Figure 21**. Note that the Called Number entry in this case represents the common matching portion applicable to all incoming numbers. Thus 0300732626 matches all numbers in the assigned DID block from BandTel.

```
change inc-call-handling-trmt trunk-group 2                    Page   1 of  30

                          INCOMING CALL HANDLING TREATMENT
 Service/       Called    Called        Del  Insert
 Feature         Len      Number
 tie              14  0300732626         9   61001
```

Figure 21: Incoming Call Handling Treatment – Simple Extension Mapping

Step 10: Save Avaya Communication Manager Changes
Enter "save translation" to make the changes permanent.

3.2 SIP Endpoint Configuration

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

Step 1: **Assign a Station**
The first step in adding an off-PBX station (OPS) for Avaya SIP telephones registered with Avaya SIP Enablement Services is to assign a station as shown in **Figure 22**.

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.

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```
add station 60000                                          Page   1 of   4
                                    STATION

Extension: 60000                         Lock Messages? n          BCC: 0
     Type: 6408D+                        Security Code:             TN: 1
     Port: X                           Coverage Path 1:            COR: 1
     Name: SipCM Phone60000            Coverage Path 2:            COS: 1
                                       Hunt-to Station:

STATION OPTIONS
             Loss Group: 2             Personalized Ringing Pattern: 1
           Data Module? n                        Message Lamp Ext: 60000
         Speakerphone: 2-way                Mute Button Enabled? y
       Display Language: english

                                                 Media Complex Ext:
                                                      IP SoftPhone? n
```

Figure 22: Station Administration – Page 1

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

- Set the **Restrict Last Appearance** value to 'n' on phones that have 3 or fewer call appearances to maintain proper SIP conference and transfer operation.
  Setting the **Restrict Last Appearance** value to 'y' reserves the last call appearance for outbound calls. Certain SIP conference and transfer features will not function properly if a third appearance is not available for incoming calls.

```
add station 60000                                          Page   2 of   4
                                    STATION
FEATURE OPTIONS
          LWC Reception: spe             Auto Select Any Idle Appearance? n
         LWC Activation? y                     Coverage Msg Retrieval? y
  LWC Log External Calls? n                          Auto Answer: none
            CDR Privacy? n                          Data Restriction? n
   Redirect Notification? y                 Idle Appearance Preference? n
 Per Button Ring Control? n               Bridged Idle Line Preference? n
   Bridged Call Alerting? n                 Restrict Last Appearance? n
  Active Station Ringing: single          Conf/Trans on Primary Appearance? n

       H.320 Conversion? n          Per Station CPN - Send Calling Number? y
      Service Link Mode: as-needed
        Multimedia Mode: basic
                                           Display Client Redirection? n
             AUDIX Name:                  Select Last Used Appearance? n
                                            Coverage After Forwarding? s
                                              Multimedia Early Answer? n
                                         Direct IP-IP Audio Connections? y
 Emergency Location Ext: 60000                     IP Audio Hairpinning? n
```

Figure 23: Station Administration – Page 2

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On Page 3 of the **Station** form, configure at least 3 call appearances under the Button Assignments section for the SIP telephone as shown in **Figure 24**.

```
add station 60000                                          Page   3 of   4
                                    STATION
 SITE DATA
      Room:                                            Headset? n
      Jack:                                            Speaker? n
     Cable:                                           Mounting: d
     Floor:                                         Cord Length: 0
   Building:                                          Set Color:



ABBREVIATED DIALING
     List1:                    List2:                     List3:

BUTTON ASSIGNMENTS
 1: call-appr                            5:
 2: call-appr                            6:
 3: call-appr                            7:
 4:                                      8:
```

Figure 24: Station Administration – Page 3

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

Step 2: **Configure Off-PBX 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 25**:

- 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 (Administration Without Hardware) stations on Avaya Communication Manager. However, this is not a requirement.
- Set the **Trunk Selection** field to *2*, which is the number assigned to the *inbound* SIP trunk group used to route the call to the SIP station. This trunk group number was previously defined in **Figure 13**.
- Set the **Configuration Set** value. In these Application Notes, Configuration Set 1 uses the default values of the Configuration Set form.

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```
change off-pbx-telephone station-mapping 60000                  Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station     Application   Dial   Phone Number      Trunk       Configuration
 Extension                 Prefix                  Selection   Set
 60000         OPS             - 60000             2           1
```

Figure 25: Stations with Off-PBX Telephone Integration – Page 1

On Page 2, set the **Call Limit** field to the maximum number of calls that may be active simultaneously at the station. In this example, the call limit is set to '3', which corresponds to the number of call appearances configured on the station form. Accept the default values for the other fields.

```
change off-pbx-telephone station-mapping 60000                  Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged
 Extension     Limit      Mode         Allowed     Calls
 60000         3          both           all         both
```

Figure 26: Stations with Off-PBX Telephone Integration – Page 2

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

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

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4. Configure Avaya SIP Enablement Services

This section covers the administration of Avaya SIP Enablement Services (SES). Avaya SIP Enablement Services is configured via an Internet browser using the Administration web interface. It is assumed that Avaya SIP Enablement Services software and the license file have already been installed on Avaya SIP Enablement Services. During the software installation, the **initial_setup** 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 three parts: **Section 4.1** provides the steps necessary to configure the Avaya SES's Primary and Secondary DNS servers which are used for resolving the Fully Qualified domain Name (FQDN) of the BandTel SIP proxy server(s). **Section 4.2** provides the steps necessary to configure a SIP trunk to BandTel's Global SIP Trunking service. **Section 4.3** provides the steps necessary to complete the administration for optional SIP endpoints.

4.1 Configure the SES's DNS Server to Resolve BandTel's SIP Proxy

BandTel provides an FQDN for their SIP proxy servers. In order for the Avaya SES to resolve this FQDN, it must query a BandTel DNS server. BandTel provides two DNS server IP addresses to the subscriber at the time of service subscription; (see **Section 5** of this document for a detailed explanation regarding the information needed and the steps necessary to setup service between BandTel and a Global SIP Trunking subscriber).

The IP address of BandTel's DNS servers must be configured as the primary and secondary DNS servers on the Avaya SES. The **initial_setup** script must be run from the SES command line interface to enter the new DNS server IP addresses. This script is typically invoked at the time of the SES's installation. For the purposes of this document, it is assumed that this was already done once and now needs to be run again in order to add the DNS configuration described above only. Other settings shown in the screens below are explained in reference [4] of this document.

Caution: The completion of the initial_setup script requires the SES be restarted. This will disrupt service to active SES subscribers.

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Step 1: Log in to the Avaya SIP Enablement Service Linux shell
Telnet or SSH to the IP address of the SES server and log in with the appropriate credentials:

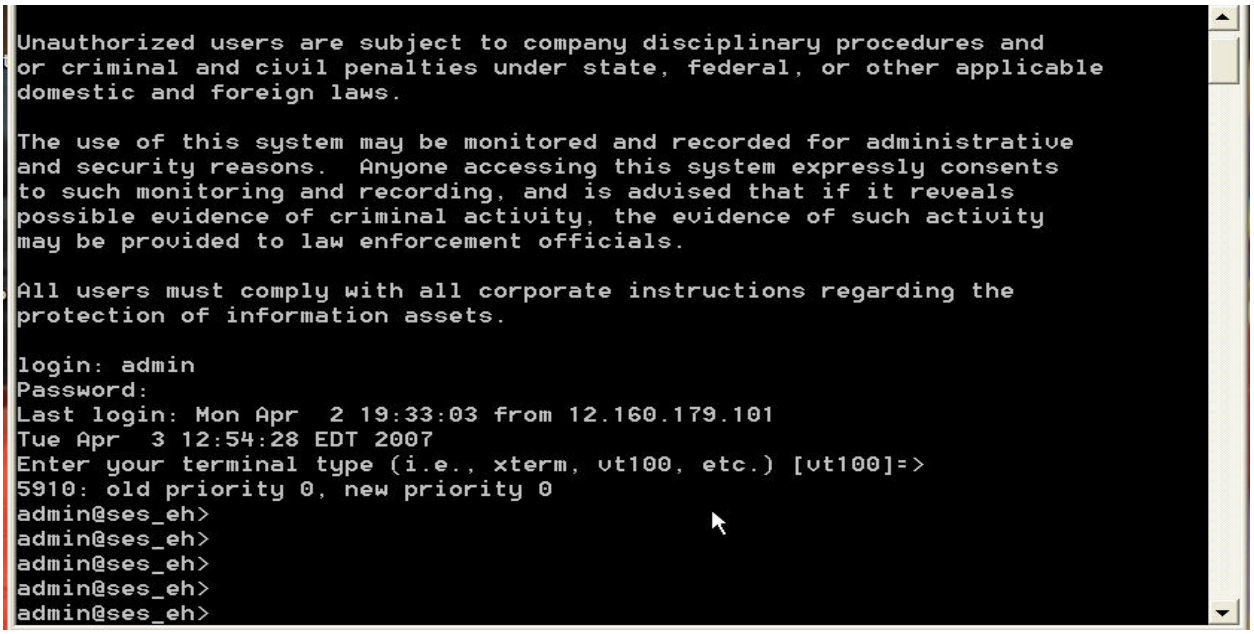


Figure 27 - Avaya SES Linux Shell

Step 2: Run the "initial_setup" script
From the shell, type **initial_setup** and press the enter key as shown in **Figure 28** below.

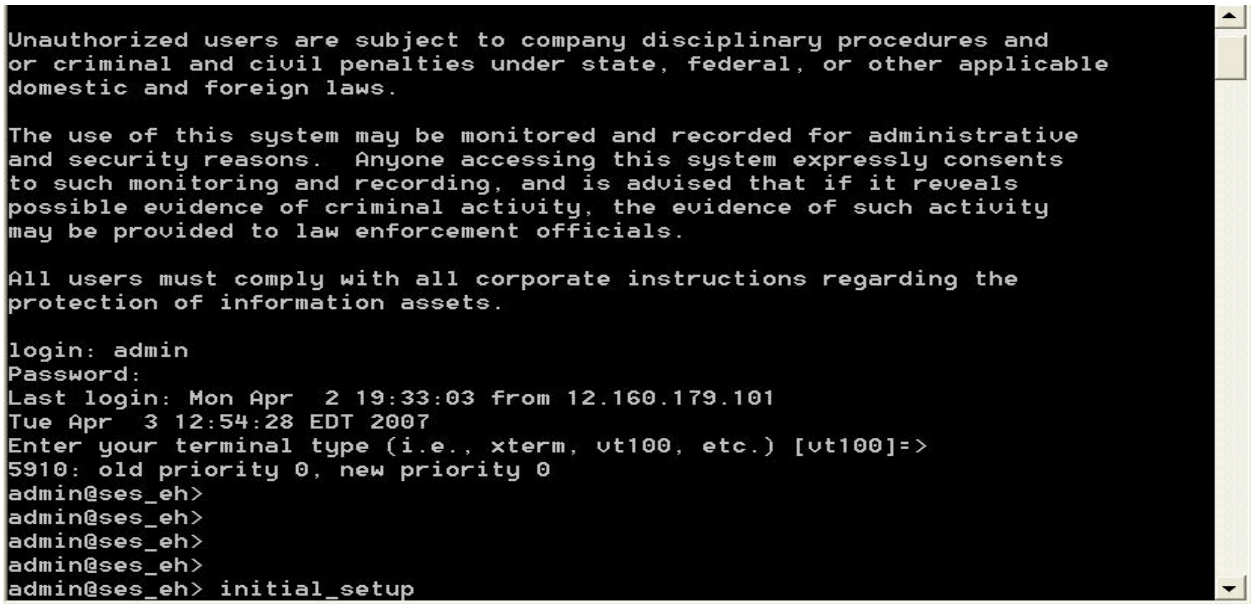


Figure 28 – Invoke initial_setup Script

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Step 3: Enter the admin password
The script will prompt the user for the admin password. After entering the password, select **OK**.



Figure 29 – Set Admin Password Screen

Step 4: Enter the Primary and Secondary DNS servers
The IP Network Configuration screen appears. In this screen, set the **Primary** and **Secondary DNS IP Address** fields with the DNS server IP addresses provided by BandTel. After doing so, select **OK**.

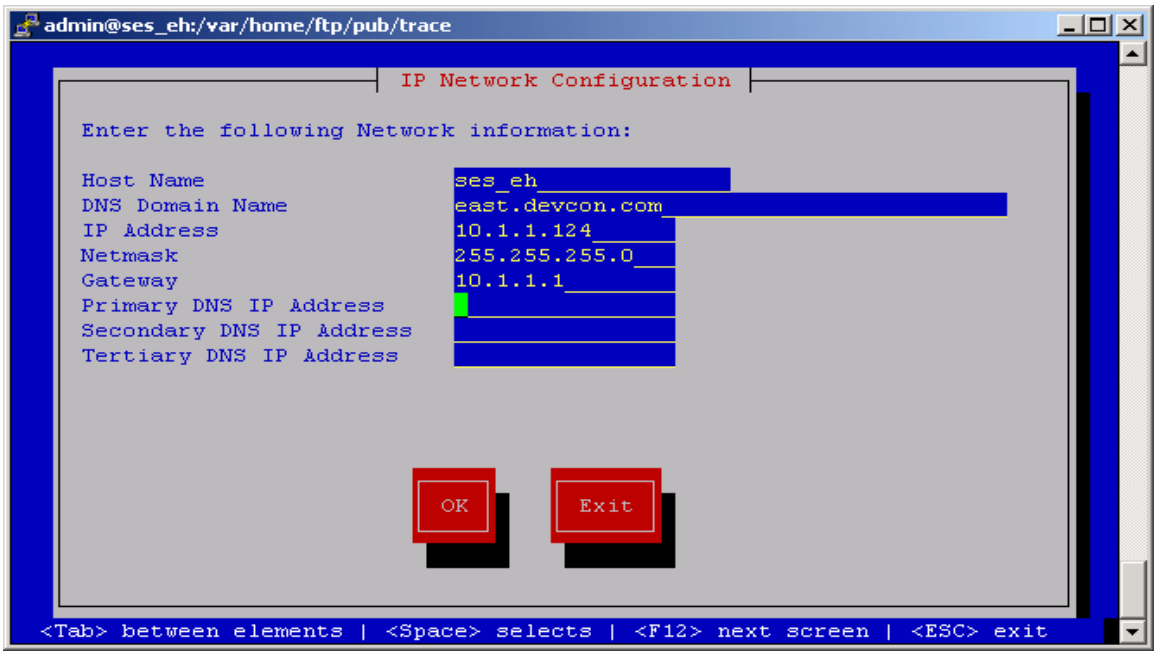


Figure 30 – IP Network Configuration Screen

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Step 5: The Redundancy Configuration screen
Assuming this was done at the time of the SES installation, simply verify the proper setting for the system and select **OK**.

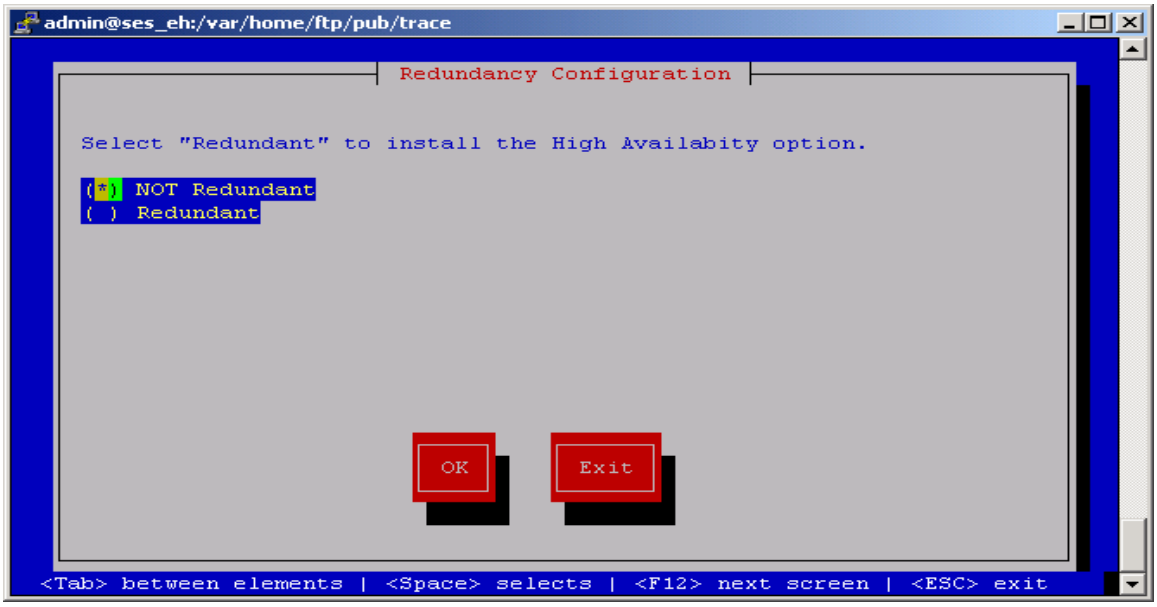


Figure 31 – Redundancy Configuration Screen

Step 6: Finish the initial script
Select **Finish** to complete the script. The system will now restart.

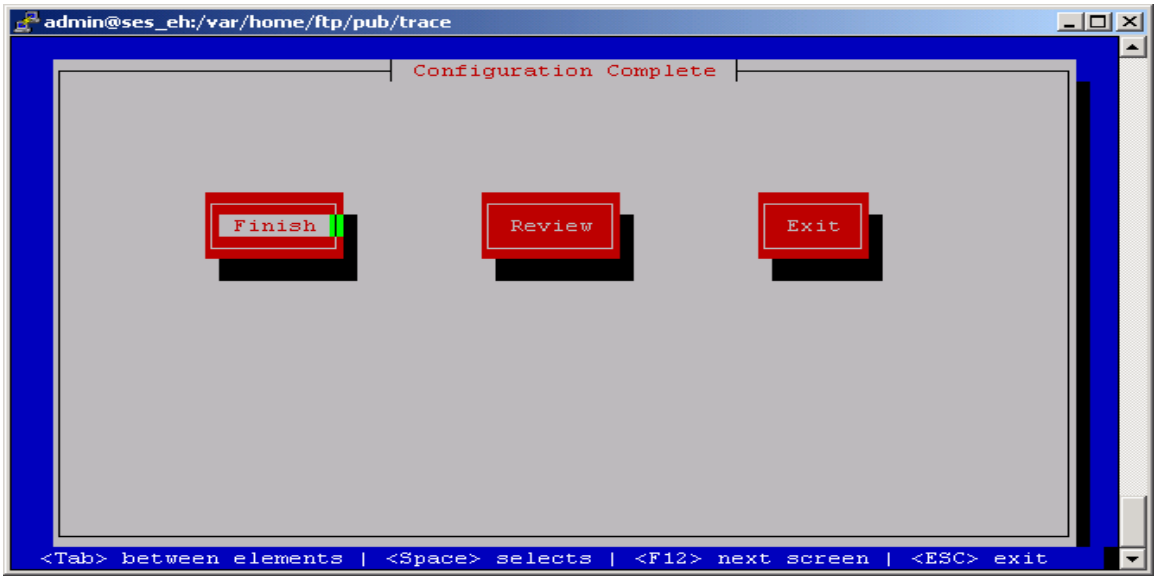


Figure 32 – Configuration Complete Screen

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4.2 SIP Trunking to BandTel

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 33**.

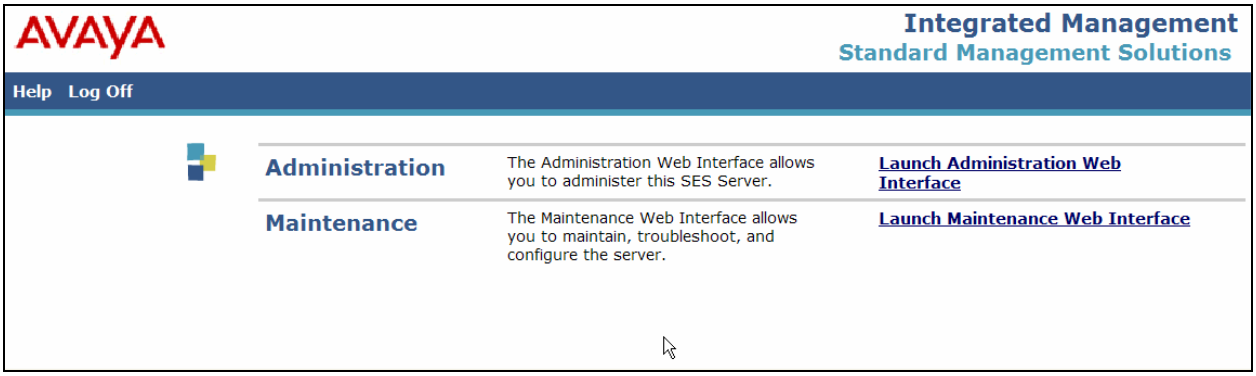


Figure 33 - Avaya SES Main Screen

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The SES administration home screen shown in **Figure 34** will be displayed.

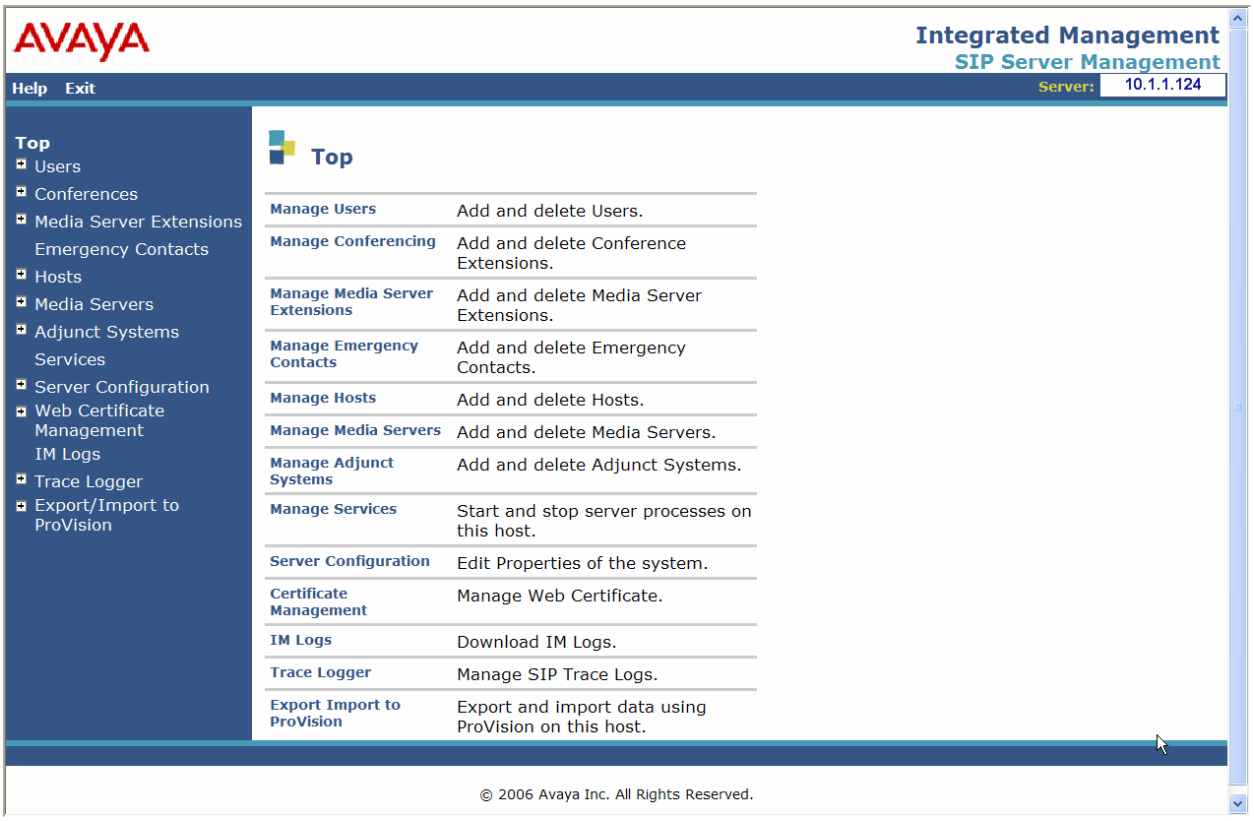


Figure 34: Avaya SES Administration Home Page

Step 2: Verify System Properties

From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. This screen displays the SES version and the network properties entered via the **initial_setup** script during the installation process.

In the **System Properties** screen,

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

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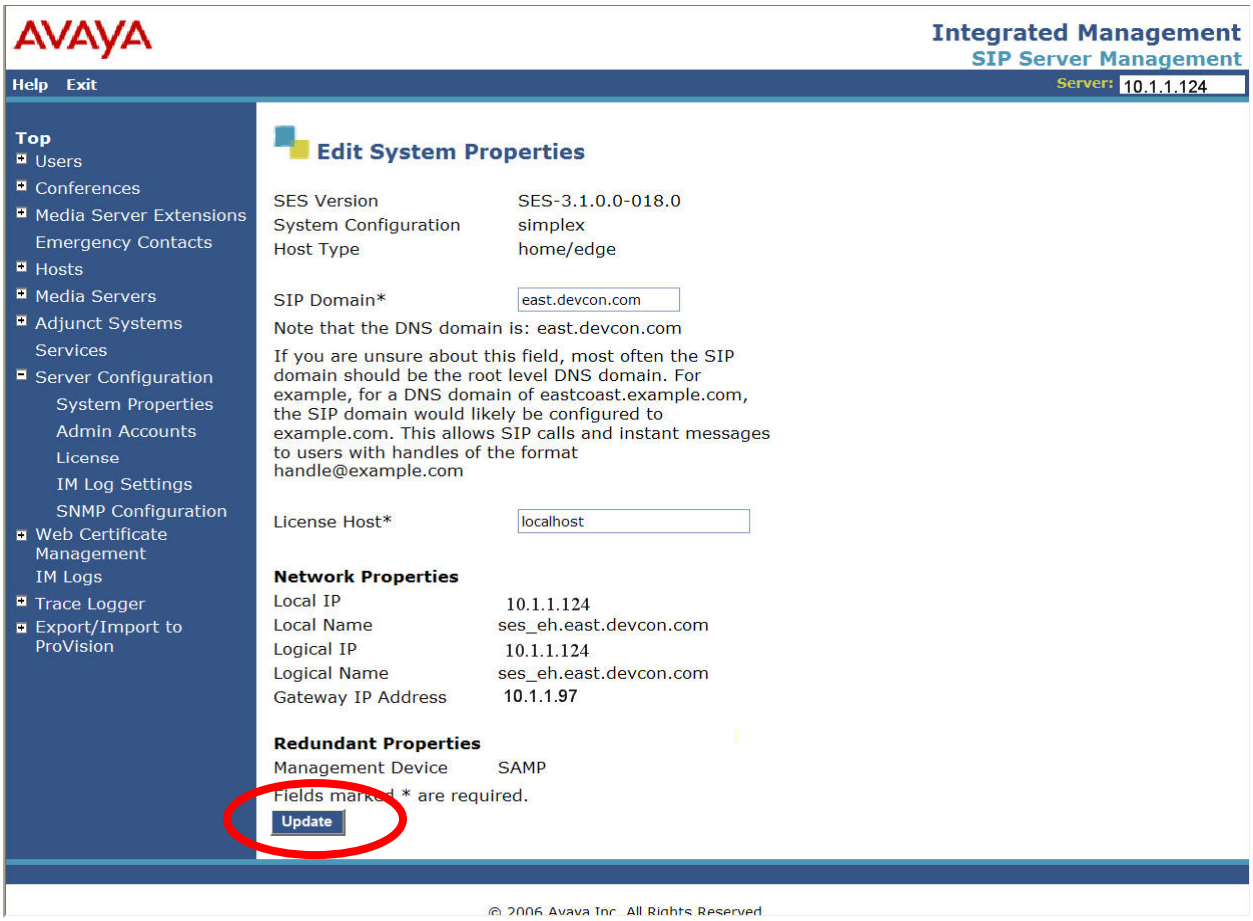


Figure 35: System Properties

Step 3: Verify the Avaya SES Host Information

Verify the Avaya SES Host information using the **Edit Host** page. In these Application Notes the Avaya SES **Host Type** is a combined ***home/edge***. This means that the both the BandTel Global SIP Trunking Service and Avaya Communication Manager are contacting the same SES. Display the **Edit Host** page (**Figure 36)** 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:

- Verify that the IP address of this combined SES Home/Edge server is in the **Host IP Address** field.
- Do not modify the **DB Password** or **Profile Service Password** fields. If these fields are changed, exit the form without using the **Update** button. These values must match the values entered during the SES installation; incorrect changes may disable the SES.
- Verify that the **UDP**, **TCP** and **TLS** checkboxes are enabled as **Listen Protocols**.
- Verify that **TLS** is selected as the **Link Protocol**.
- Default values for the remaining fields may be used.
- Click the **Update** button only if changes are necessary. Otherwise exit the **Edit Host** page by selecting the **Top** link on the left navigation bar.

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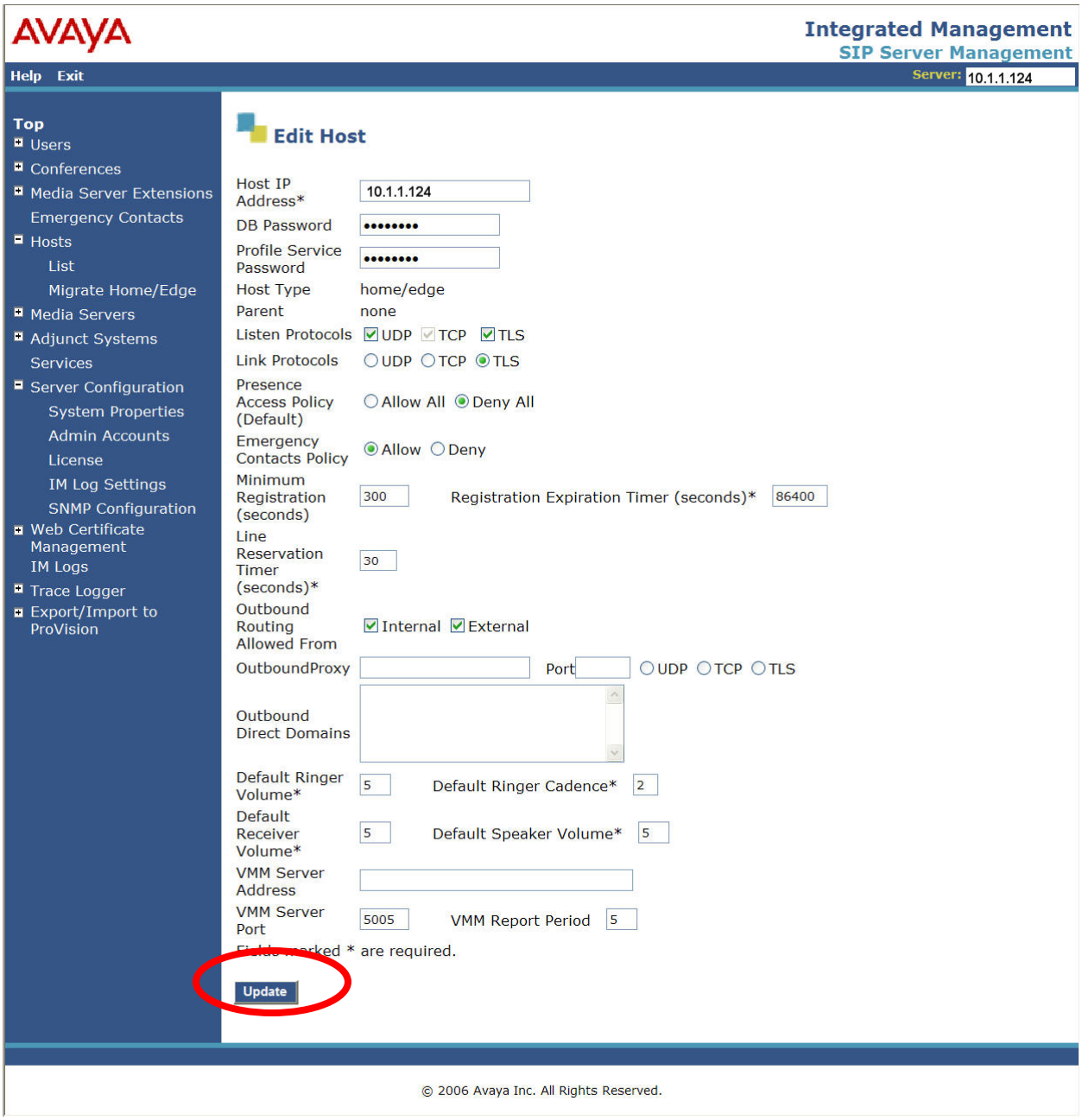


Figure 36: Edit Host

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-SignalGroup1).
- Select IP address of the home SES server in the **Host** field as specified in **Figure 36**.
- 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 such as an Avaya S8300 Media Server using an Avaya G350 Media Gateway.)
- After completing the **Add Media Server** screen, click on the **Add** button.

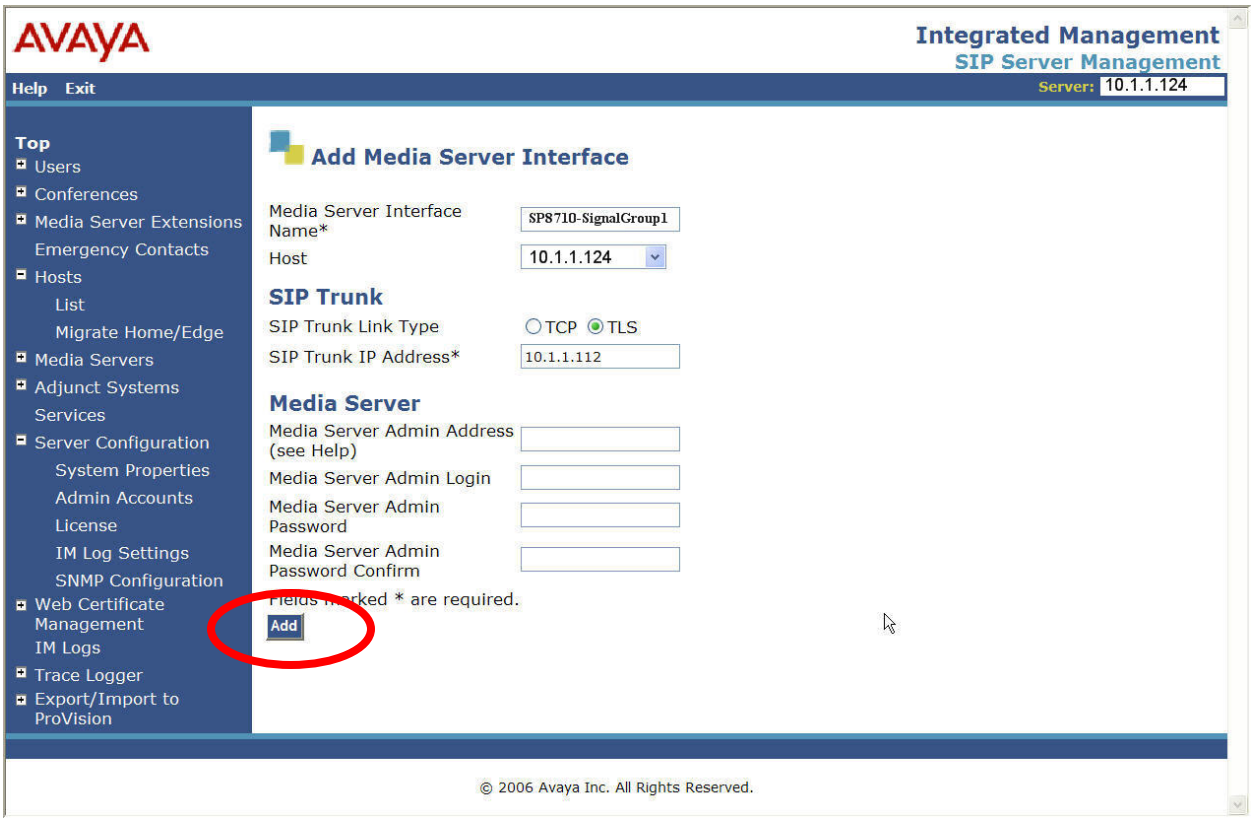


Figure 37: Add Media Server

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

This routing compares the Uniform Resource Identifier (URI) of an incoming INVITE message to the pattern configured in the Media Server Address Map**,** and if there is a match, the call is routed to the designated Avaya Communication Manager. The URI usually takes the form of *sip:user@domain*, where *domain* can be a domain name or an IP address. Patterns must be specific enough to uniquely route incoming calls to the proper destination if there are multiple Avaya Communication 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 BandTel's Global SIP Trunking service, the *user* portion of the SIP URI will contain the 14 digit value specified for the incoming direct inward dialed telephone number. An example of a SIP URI in an INVITE message received from BandTel would be:

sip:03007326269601@10.1.1.124;user=phone;npdi=yes

Note: *The **npdi=yes** field refers to the Number Portability Dip Indicator and simply implies that a dip was made into the number portability data base upstream.*

The user portion in this case is the 14 digit DID number "03007326269601". 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 BandTel. 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 38** is displayed. The **Host** field displays the name of the media server that this map applies to.
- Enter a descriptive name in the **Name** field
- Enter the regular expression to be used for the pattern matching in the **Pattern** field. In this configuration, the DID numbers provided by BandTel are 732-626-9601 thru 9604 with a "0300" prefix. ***(Note: The full string is not visible in Figure 38 due to the text box size)*** The pattern specification (without the double quotes) for DID numbers assigned is:

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"^sip:0300732626960[1234]". This means that URIs beginning with "sip: "0300732626960" followed by the digits 1, 2, 3 and 4 will match the pattern and be routed to the interface defined for S8710-CLAN. Appendix B provides a detailed description of the syntax for address map patterns.

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

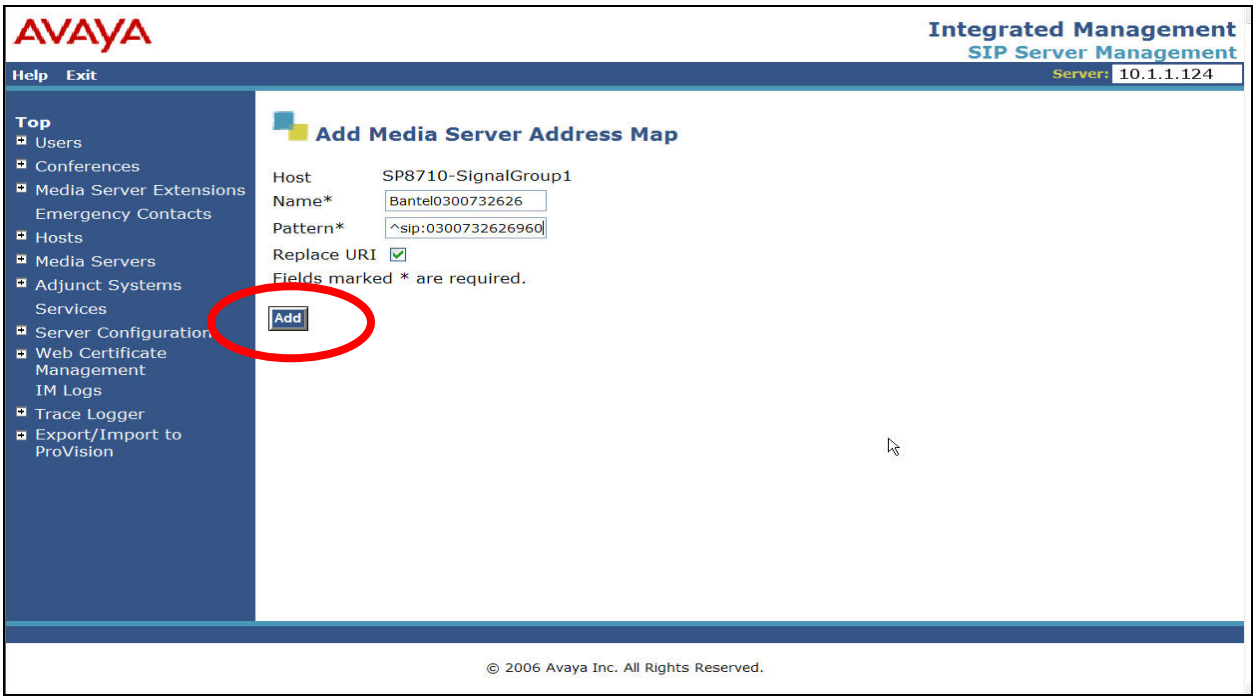


Figure 38: Add Media Server Address Map

After configuring the media server address map, the **List Media Server Address Map** screen appears as shown in **Figure 39**.



Figure 39: List Media Server Address Map

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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 38**, the following contact was created:

sips:$(user)@10.1.1.112:5061;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 BandTel

Outbound PSTN calls are directed by Avaya 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 BandTel's Global SIP Trunking 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 BandTel SIP proxy 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 BandTel SIP Proxy). In this configuration, the Avaya SES routing rule for the SIP trunk group will be to send all outbound PSTN traffic to BandTel's Global SIP Trunking 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.

In addition, the following maps could be added should the dial plan include operator and N11 service:

- 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.

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The configuration of the host address map for all 1+ North America calls is shown in **Figure 34**.

- 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. 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 40**. Enter a descriptive name for the map, such as "BandTel_Out_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

Additional Host Address Map patterns can be added in a similar manner.

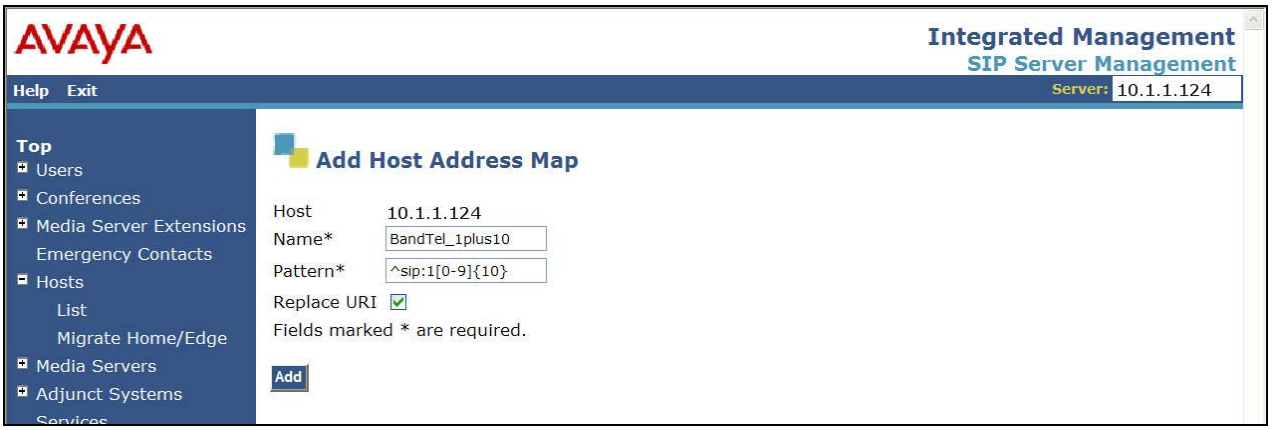


Figure 40: Add Host Address Map

Step 7: Specify the BandTel SIP Proxy Information

The next step is to enter the contact address for the BandTel SIP gateway. In this example, a DNS name is used to identify BandTel's SIP proxy. This customer specific information is provided by BandTel.

To enter the BandTel SIP proxy information:

- As described in **Step 6**, display the **List Host Address Map** screen.
- Click on the **Add Another Contact** link associated with the address map added in **Figure 40** 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)@proxyX.bandtel: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 BandTel. The FQDN "proxyX.bandtel" is resolved via a query to BandTel's DNS servers, see **section 4.1** of this document for a detailed explanation.
- Click the **Add** button when completed.

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After configuring the host address maps and contact information, the **List Host Address Map** screen will appear as shown in **Figure 41**.

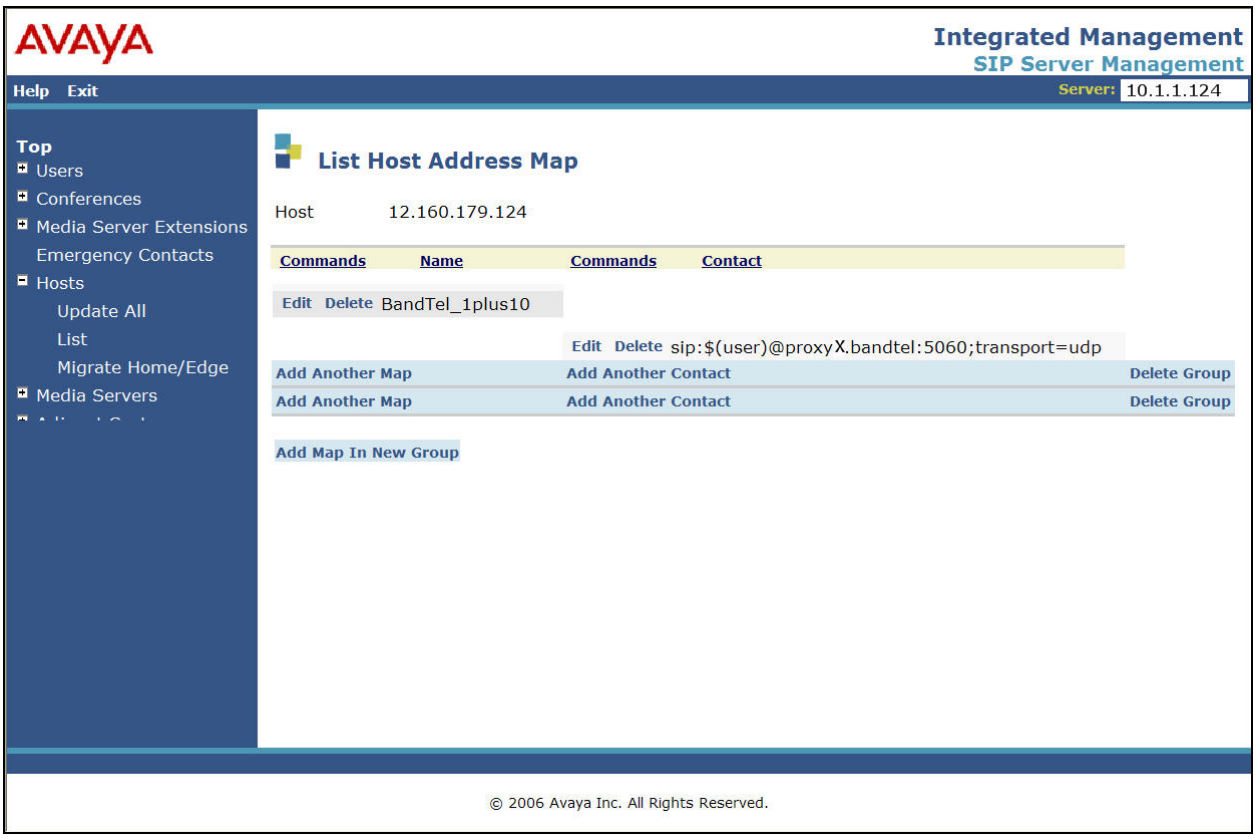


Figure 41: List Host Address Map

Step 8: Save the Changes

After making changes within Avaya SES, it is necessary to commit the database changes using the **Update** link that appears when changes are pending. Perform this step by clicking on the **Update** link found in the bottom of the blue navigation bar on the left side of any of the SES Administration screens as shown in **Figure 42**.

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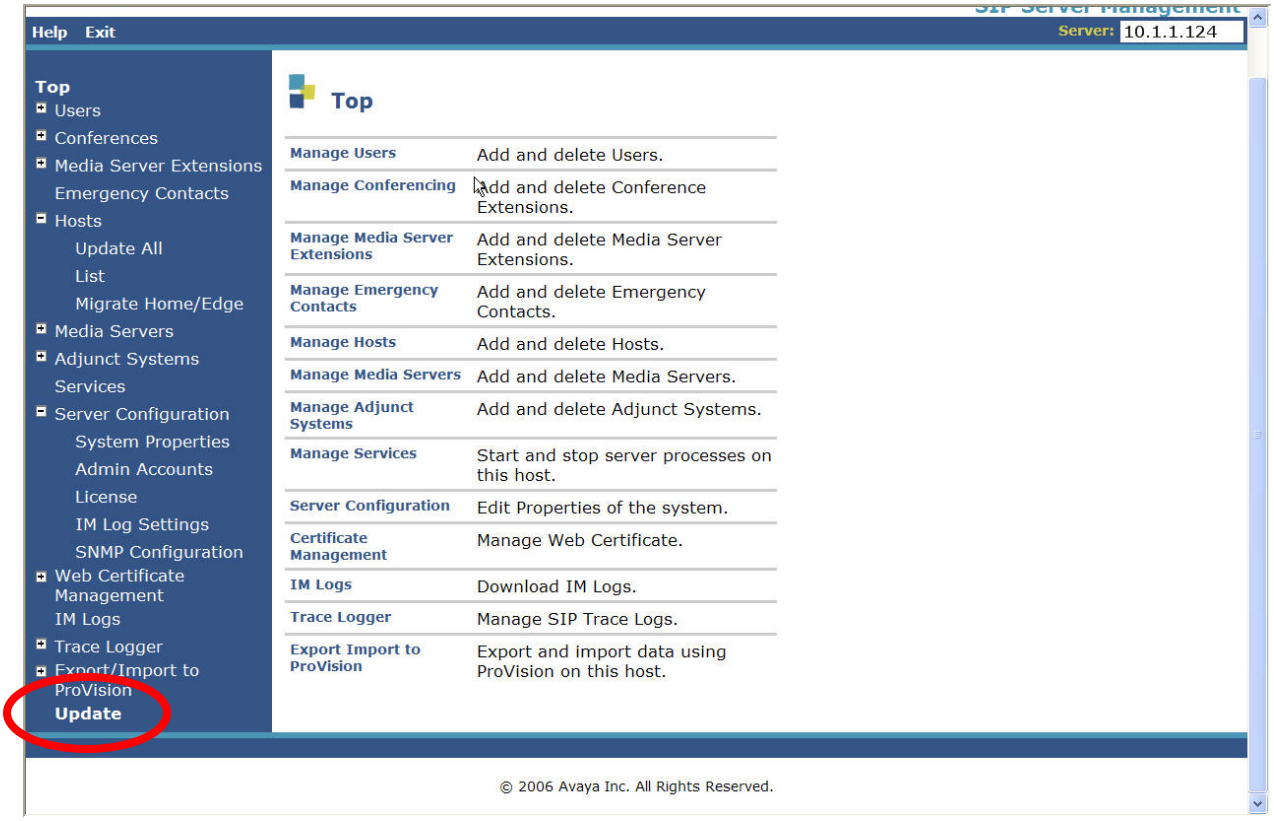


Figure 42: Update Following SES Administrative Changes

Step 9: Specify the BandTel SIP Proxy as a Trusted Host

The final step to complete the SIP trunk administration on Avaya SES is to designate the IP address of BandTel 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. For a BandTel configuration, a FQDN name is entered instead of the IP address. The Trusted Host process uses the SES's DNS process to resolve the FQDN name.

[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.

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To configure a trusted host:

- Telnet or SSH to the Avaya SES Linux shell using the administrative login and password.
- Enter the following trusted host command at the Linux shell prompt:

  trustedhost –a proxyX.bandtel –n 10.1.1.124 –c Bandtel

  The –a argument specifies the address to be trusted; –n specifies the SES host name; –c adds a comment. Once the entry is made, a DNS lookup is automatically invoked to resolve the FQDN. The resolved address is held in an expanded table not visible to the user.

- Use the following trusted host command to verify the entry is correct:

  trustedhost –L

Figure 43 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 42**.

  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.

```
admin@ses_eh> trustedhost -L
Third party trusted hosts.
       Trusted Host       |       CCS Host Name       |         Comment
--------------------------+---------------------------+-----------------------
proxyX.bandtel            | 10.1  .1  .124            | BandTel
admin@ses_eh>
```

Figure 43: Configuring a Trusted Host

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

trustedhost –d proxyX.bandtel –n 10.1.1.124

removes the trust relationship added above.

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4.3 Configuration for SIP Telephones

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

Step 1: Add a SIP User

Create the SIP user record as follows:

- In Avaya SES administration, expand the **Users** link in the left side blue navigation bar and click on the **Add** link.
- In the **Add User** screen, enter the extension of the SIP endpoint in the **Primary Handle** field.
- Enter a user password in the **Password** and **Confirm Password** fields. This password will be used when logging into the user's SIP telephone.
- In the **Host** field, select the Avaya SIP Enablement Services server hosting the domain (*10.1.1.124*) for this user. Enter the **First Name** and **Last Name** of the user.
- To associate a media server extension with this user, select the **Add Media Server Extension** checkbox. Calls from this user will always be routed through Avaya Communication Manager over the SIP trunk for origination services.
- Press the **Add** button. This will cause a confirmation screen to appear.
- Press **Continue** on the confirmation screen.

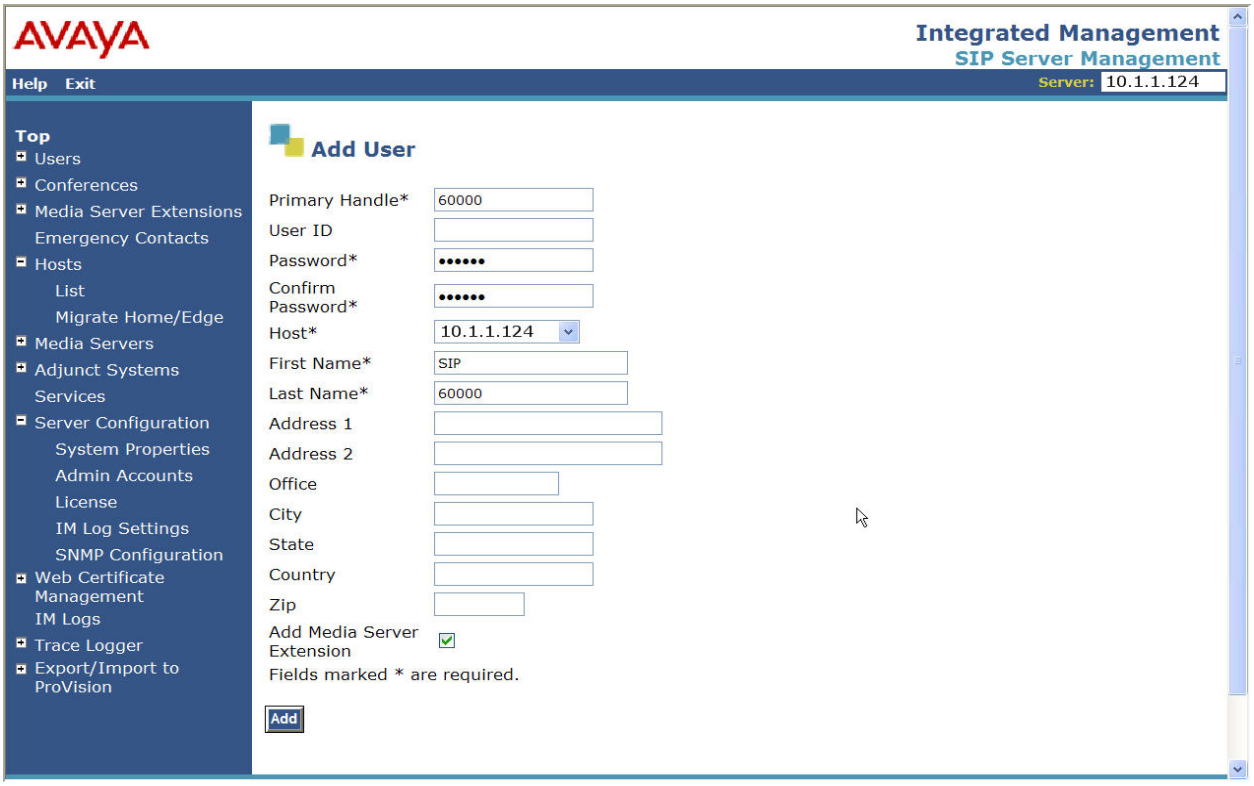


Figure 44: Add User

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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 22**, 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.
- Select the **Media Server** assigned to this extension.
- Click on the **Add** button.

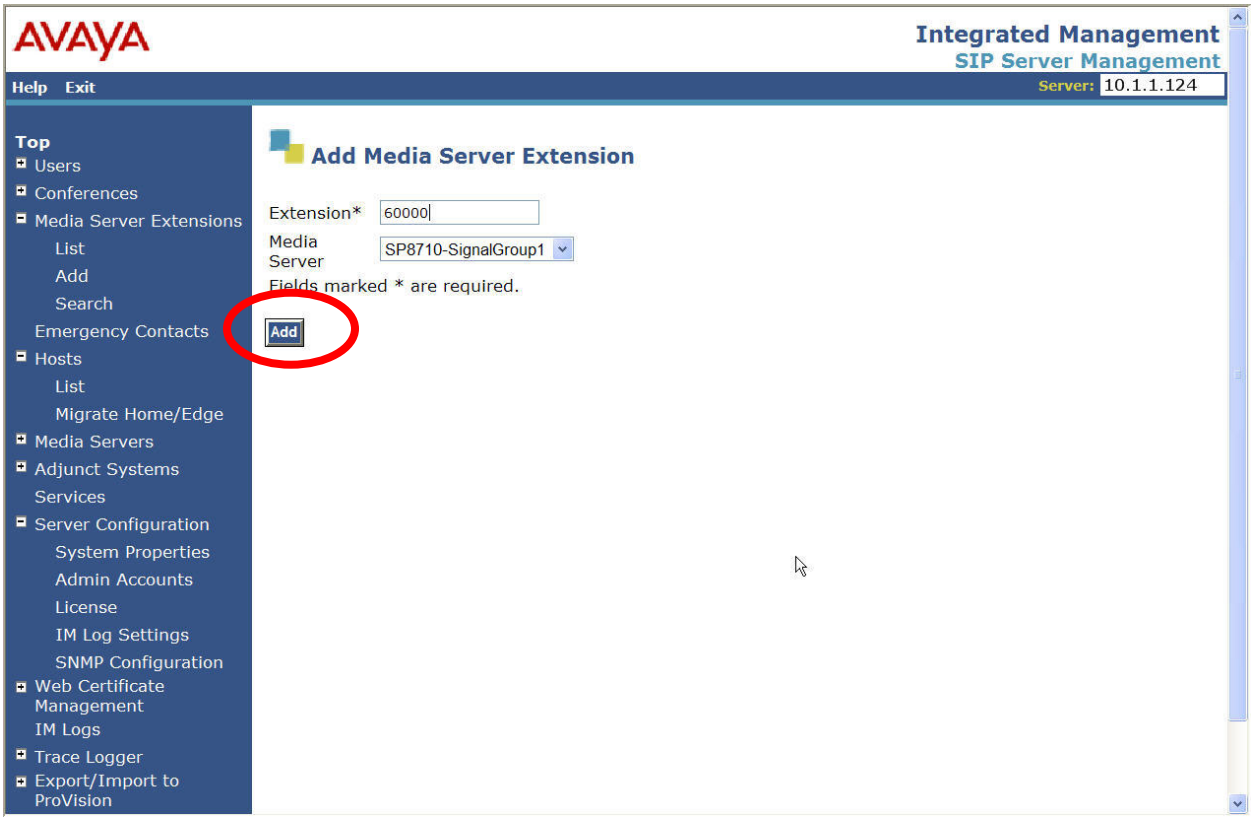


Figure 45: Add Media Server Extension

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

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5. BandTel SIP Trunking Services Configuration

In order to use BandTel's Global SIP Trunking service, a customer must request service from BandTel using their sales process. The process can be started by contacting BandTel via their corporate web site at http://www.bandtel.com/contact.asp and requesting information via the online sales links or telephone numbers.

During the signup process, BandTel will require that the customer provide the public IP address used to reach the Avaya SIP Enablement Services. (Note the address used within these Application Notes is 10.1.1.124; the actual IP address will be specific to the customer implementation).

BandTel states that the following are mandatory requirements:

- SIP V2.0 Compliant
  → **Avaya's SIP Solution is V2.0 Compliant.**

- Proxy By DNS Naming
  → **For outbound calls, BandTel requires that the customer gateway communicate with a DNS name representing BandTel's SIP proxy server rather than an IP address. This is due to their network architecture which load balances between multiple parallel proxy servers. In order for the SES to resolve this proxy server's DNS name, the SES must use BandTel's DNS servers. BandTel will give the customer two DNS servers that the user must set to the Primary and Secondary DNS server entries in the SES via the "initial_setup" script.**

- Redundant DNS Servers
  → **The Avaya SES is capable of querying two DNS servers.**

- SIP 302 REDIRECT
  → **Avaya's SIP solution does not currently support this feature. This must be stated to your BandTel contact in order to work around this limitation.**

- IP Authentication
  → **For SIP registration and authorization, BandTel typically prefers that the customer's SIP proxy respond to BandTel's dynamic challenge/answer model. Since SES does not support this model, it is necessary to request that BandTel automatically IP Authenticate our SES. Essentially, they will hard code the SES's IP address into their IP authenticated database.**

Following signup, BandTel will provide the following:

- IP address of BandTel DNS servers
- Two Fully Qualified Domain Name for two SIP proxies
- Direct Inward Dialed (DID) Numbers

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

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6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between BandTel's Global SIP Trunking 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 consisting of an Avaya IP telephony solution supporting SIP trunking was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the commercially available Global SIP Trunking Service provided by BandTel. 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 BandTel.
- Outgoing calls from the enterprise site were completed via BandTel to the PSTN destinations.
- Calls using SIP, H.323, digital and analog endpoints supported by the Avaya IP telephony solution.
- Various call types including: local, long distance, international, toll free, and directory assistance calls.
- Calls using G.711mu codec and G.729.
- Fax routing to ensure G.711mu use for fax calls.
- DTMF tone transmission using RFC 2833 with successful Voice Mail/Vector navigation
- Telephone features such as hold, transfer, conference.
- Direct IP-to-IP media (also known as "shuffling") with SIP/H323 telephones.

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6.2. Test Results

Interoperability testing of the sample configuration was completed with successful results.

The following minor issues described below were observed.

6.2.1 RFC 2833 Telephone Event Value 127

Issue Observed

Outbound calls may or may not use RFC 2833 for DTMF transmission when the default Avaya telephone event value of 127 is used. Inbound calls will use RFC 2833 when G.711mu law or G.729b is used for the codec type.

Discussion/Workaround

Outbound Calls:
The value 127 is valid according to RFC 2833, see reference [8].

However, BandTel operates as a SIP Back-to-Back User Agent. This means that BandTel does not negotiate key parameters of the placed call including the telephone event or codec value. Instead, the call is routed to one of numerous carriers with which BandTel has a service relationship. It is one of these carriers that negotiate the telephone event and codec value.

According to BandTel, some of the carriers do not support the value of 127 for RFC 2833. Therefore, the use of RFC 2833 for DTMF transmission would be prevented.

Inbound Calls:
BandTel offers the telephone event value of 100. The Avaya SIP Solution accepts this value and RFC 2833 works when G.711mu codec or G.729b is used.

Note: Avaya Communication Manager release R3.1.3 and R4.0, this problem no longer exists due to the added capability to modify the Telephone Event Value. The administrator can set this value to match the value used by the Service Provider's network.

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6.2.2 IP- IP Direct Audio

Issue Observed

IP – IP Direct Audio is not supported via BandTel's network for outbound or inbound calls.

Discussion/Workaround

Inbound & Outbound Calls:
In order to activate the IP – IP Direct Audio feature, the existing SIP session must be modified. There are two methods for modifying an existing SIP session according to RFC 3261:

1) Send the modified parameters in a Re-Invite via SDP.
2) Send the Re-Invite without the modified parameters i.e. no SDP, then wait for the 200 level responses to contain the SDP with modified parameters.

The Avaya SIP Solution does this by using method two.

According to BandTel, due to limitations of the carrier SIP equipment, method two is not supported. Therefore, IP – IP Direct Audio must be disabled in Avaya Communication Manager for the signaling group used for the BandTel service using the Signaling-Group form.

The impact of this limitation is related to efficient use of hardware resources. When the Direct IP feature is not used, voice/media traffic must travel through a hardware module on the Avaya G650 Media Gateway. This module has finite resources and therefore it is more efficient to free this resource by using the Direct IP feature.

6.2.3 Codec Support

Issue Observed

G.711mu law is the primary codec fully supported by the BandTel service. A request to use the G.729b codec made to BandTel on a customer by customer basis.

Discussion/Workaround

BandTel operates as a SIP Back-to-Back User Agent. This means that BandTel does not negotiate key parameters of the placed call including the telephone event or codec value.

Instead:
Outbound calls from Avaya's SIP Solution to BandTel's Network are routed to one of numerous carriers with which BandTel has a service relationship. It is one of these carriers that negotiate the codec value. Based upon the agreement between BandTel and the carrier, the codec value is G.711mu.

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Inbound calls from the carrier across BandTel's network and to Avaya's SIP Solution already have the codec offered set to G.711mu by the carrier. Again, this is based upon the agreement between that carrier and BandTel. G.729b can be offered by a special request but is NOT recommended according to BandTel.

6.2.4 Improper Disconnect When Calling Party Hangs Up Before Call is Answered

Issue Observed

If a call is placed from an Avaya based solution into the BandTel Network and then the call is canceled before the far end phone is picked up, BandTel fails to disconnect the call properly. This results in a protocol timeout issue.

Discussion/Workaround

No work around is needed. This issue is transparent to the end user; it is a minor protocol issue.

BandTel is in the process of fixing this.

When the calling party disconnects before the called party answers, a SIP Cancel is sent to BandTel by the Avaya SES, in turn, BandTel should generate a "487 Request to Terminate" message back to Avaya. When they do this, they fail to include a TAG in the "To:" header field of this message, as a result, the Avaya SES proxy does not associate this 487 with the previous transaction. This results in miscommunication between BandTel and Avaya. BandTel continues to send these 487 Request To Terminate messages until it reaches a time out threshold.

This does not affect the tear down of the call.

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7. Verification Steps

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

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.

8. Support

For technical support on BandTel's Global SIP Trunking Service, contact BandTel Customer Service at 1-800-765-7069 or 603-528-6538 or email support requests to support@bandtel.com

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 BandTel's Global SIP Trunking service. BandTel's Global SIP Trunking Service is a robust Voice over IP solution for customers ranging from small businesses to large enterprises. SIP trunks use the Session Initiation Protocol (SIP) 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.

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10. References

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

[1] *Administrator Guide for Avaya Communication Manager*, May 2006, Issue 2.1, Document Number 03-300509.

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

[3] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.1*, Feb 2006, Issue 9, Document Number 210-100-700.

[4] *Installing and Administering SIP Enablement Services R3.1*, February 2006, Issue 1.5, Document Number 03-600768

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

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

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

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

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APPENDIX A: Sample SIP INVITE Messages

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

Note: For the incoming BandTel Invite message, *"proxyX.bandtel"* is used in the Invite below in place of the actual IP address of the BandTel SIP Proxy for security purposes.

INVITE sip:03007326269601@10.1.1.124 SIP/2.0
Record-Route: <sip:*proxyX.bandtel*;ftag=ef87d6205aeabae9bd125be903323811;lr>
Via: SIP/2.0/UDP *proxyX.bandtel*;branch=z9hG4bKdc8.47076a99ec7c7e1989b71198e2a2c0f9.0
Via: SIP/2.0/UDP
bandtel-proxy:5061;branch=z9hG4bK924fdc4f2852f32d18834a5962557442;rport=5061
Max-Forwards: 16
From: <sip:7328521637@ *proxyX.bandtel* >;tag=ef87d6205aeabae9bd125be903323811
To: <sip:03007326269601@ *proxyX.bandtel* >
Call-ID: SDtin9601-09c7a5bcf5abeac053f0819e393ab06c-v3000i1
CSeq: 200 INVITE
Contact: Anonymous <sip: *proxyX.bandtel*:5061>
Expires: 300
User-Agent: Sippy
cisco-GUID: 2527960762-1277017105-672791513-2626401345
h323-conf-id: 2527960762-1277017105-672791513-2626401345
Portabilling-notify: aor=2043600001
P-charge-info: <sip:6035286538@ *proxyX.bandte* >;npi=ISDN
Content-Length: 258
Content-Type: application/sdp

Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): Sippy 156495948 0 IN IP4 *proxyX.bandtel*
Session Name (s): SIP Media Capabilities
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 50040 RTP/AVP 0 8 100
Connection Information (c): IN IP4 *proxyX.bandtel*
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:100 telephone-event/8000
Media Attribute (a): fmtp:100 0-15
Media Attribute (a): sendrecv
Media Attribute (a): maxptime:20

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

Note: For the outgoing Invite sent by the Avaya SES below, an FQDN is in fact used to reference the BandTel SIP proxy as discussed throughout this document. Used below is the FQDN "proxyX.bandtel".

INVITE sip:17328521637@proxyX.bandtel SIP/2.0
Call-ID: 803232d76f9adb1c922459ce6a900
CSeq: 1 INVITE
From: "Dcp Phone62004" <sip:62004@east.devcon.com:5061>;tag=803232d76f9adb1c822459ce6a900
Record-Route: <sip:10.1.1.124:5060;lr>,<sip:12.160.179.112:5061;lr;transport=tls>
To: "17328521637" <sip:17328521637@proxyX.bandtel>
Via: SIP/2.0/UDP 10.1.1.124:5060;branch=z9hG4bK838383030303333333c2af.0,SIP/2.0/TLS 12.160.179.112;psrrposn=2;branch=z9hG4bK803232d76f9adb1ca22459ce6a900
Content-Length: 158
Content-Type: application/sdp
Contact: "Dcp Phone62004" <sip:62004@10.1.1.112:5061;transport=tls>
Max-Forwards: 69
User-Agent: Avaya CM/R013x.01.2.632.1
Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
Session-Expires: 240;refresher=uac
Min-SE: 240
History-Info: <sip:17328521637@proxyX.bandtel>;index=1
History-Info: "17328521637" <sip:17328521637@proxyX.bandtel>;index=1.1
Supported: 100rel,timer,replaces,join,histinfo
P-Asserted-Identity:"Dcp Phone62004"<sip:62004@east.devcon.com:5061>

Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 1 1 IN IP4 10.1.1.112
Session Name (s): -
Connection Information (c): IN IP4 10.1.1.116
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 2524 RTP/AVP 0 127
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:127 telephone-event/8000

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APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within the Avaya SES is a Linux regular expression used to match against the URI string found in the SIP INVITE message.
Regular expressions are a way to describe text through pattern matching. The regular expression is a string containing a combination of normal text characters, which match themselves, and special *metacharacters*, which may represent items like quantity, location or types of character(s).
In the pattern matching string used in the Avaya SES:

- Normal text characters and numbers match themselves.
- Common metacharacters used are:
    - A period **.** matches any character once (and only once).
    - 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@proxy-bandtel:5060;transport=udp SIP/2.0

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