Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking Between the Tello Corporation Tello Connect Service and an Avaya SIP-Enabled IP Telephony Network - Issue 1.0

Abstract

These Application Notes describe the configuration of SIP trunking between the Tello Corporation Tello Connect service and an Avaya SIP-enabled IP telephony network comprised of Avaya SIP Enablement Services, Avaya Communication Manager, and Avaya SIP, H.323, digital and analog telephones.

Tello Connect is a hosted service that provides for the interconnection of Voice over IP (VoIP) calls across enterprise network boundaries via the Internet using the Session Initiation Protocol (SIP). The SIP signaling messages are passed through the service, but not the media stream. Once the call is established, the media stream is passed directly between the two enterprise sites via the Internet. This solution allows enterprise customers with a converged network to reduce telecom expenses by eliminating the need to route calls through traditional telephone networks. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Service Provider Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration of SIP trunking between the Tello Corporation Tello Connect service and an Avaya SIP-enabled IP telephony network comprised of Avaya SIP Enablement Services, Avaya Communication Manager, and Avaya SIP, H.323, digital and analog telephones.

Tello Connect is a hosted service that provides for the interconnection of Voice over IP (VoIP) calls across enterprise network boundaries between Tello Connect subscribers. The interconnection is done via the Internet using the Session Initiation Protocol (SIP). This reduces telecom expenses between subscribers by eliminating the need to route calls through traditional telephone networks. Individual users may not know if the number they are dialing belongs to another Tello Connect subscriber. Thus, it is the intent that all outbound calls from all users at the enterprise site be routed to Tello Connect first for call completion. Non-SIP endpoints supported by Avaya Communication Manager can also use this service. Avaya Communication Manager provides all necessary protocol conversions to allow these endpoints to access the SIP trunk that connects to the service.

Tello Connect maintains a database of service subscribers including the range of phone numbers owned by each subscriber and the SIP proxy address servicing calls to each location. When Tello Connect receives a call from a subscriber, Tello Connect will check if the destination phone number matches an entry in the database. If a match exists, then Tello Connect will forward the call to the necessary SIP proxy. If no match exists, then the call is rejected with a 406 error code. The enterprise then needs to take the necessary steps to reroute the call, presumably via the public-switched telephone network (PSTN). The SIP signaling messages are passed to Tello Connect, but not the media stream. Once the call is established, the media stream is passed directly between the two enterprise sites via the Internet.

Figure 1 illustrates the network configuration used in the compliance testing. The configuration shows two enterprise sites each of whom subscribe to Tello Connect. Tello Connect Subscriber A has an Avaya SIP-enabled IP telephony network that includes Avaya SIP Enablement Services (SES), an Avaya S8300 Media Server running Avaya Communication Manager and an Avaya G350 Media Gateway. Endpoints include Avaya 4600 Series SIP Telephones, an Avaya 4600 Series IP (H.323) Telephone, an Avaya 6400 Series Digital Telephone, and an Avaya 6200 Series Analog Telephone. Tello Connect Subscriber B has an Avaya SIP-enabled IP telephony network that includes Avaya SIP Enablement Services (SES), a redundant pair of Avaya S8710 Media Servers running Avaya Communication Manager, and an Avaya G650 Media Gateway. Endpoints include an Avaya 4600 Series SIP Telephone, an Avaya 4600 Series IP (H.323) Telephone, an Avaya 6400 Series Digital Telephone, and an Avaya 6200 Series Analog Telephone. Each enterprise site and the Tello Connect service are connected to the Internet. Each enterprise site also has an ISDN-PRI trunk connection to the PSTN.

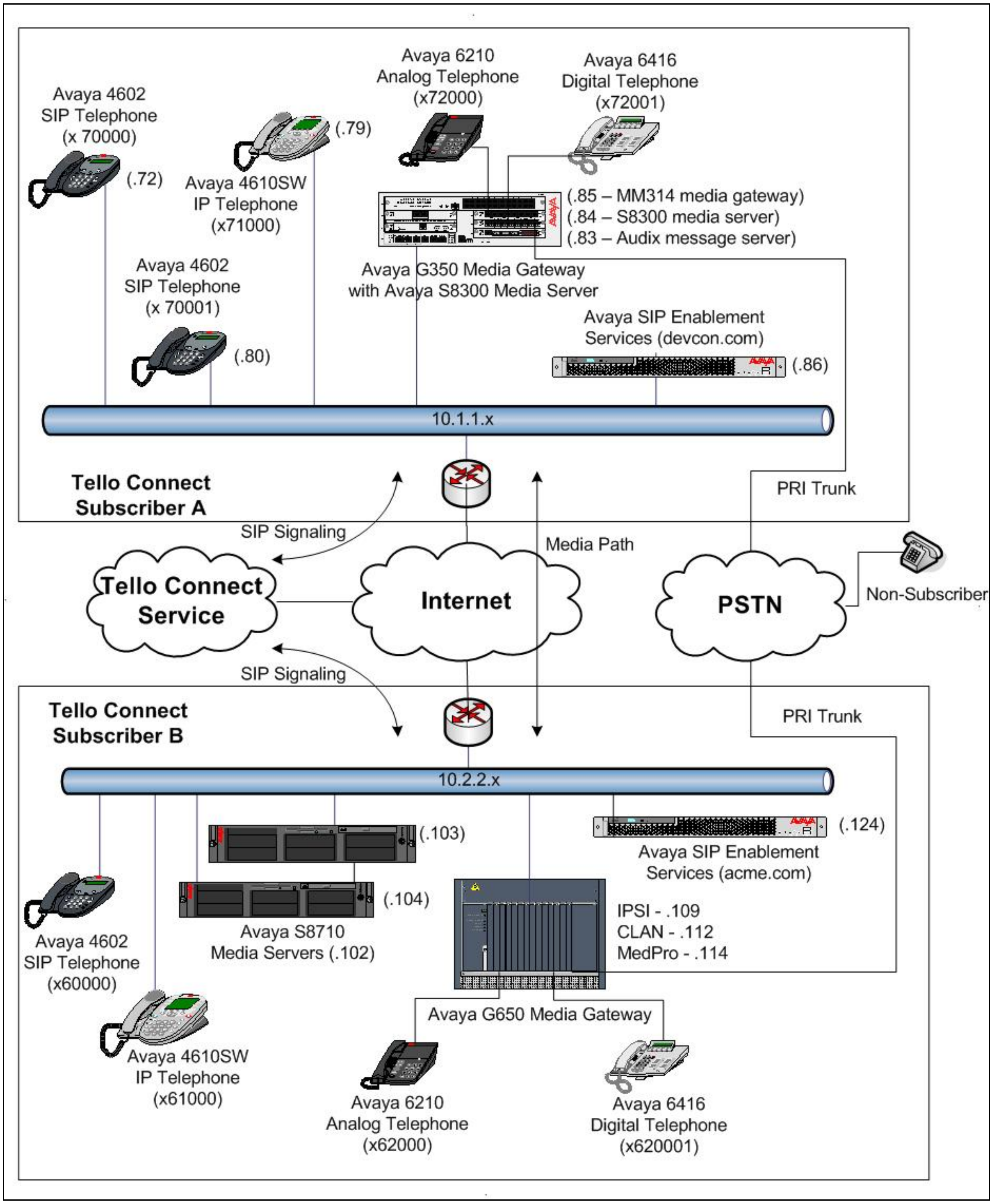

Figure 1: Two Avaya SIP-enabled IP Telephony Networks Each Subscribed to the Tello Connect Service

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

Figure 1 does not show the actual public IP addresses that were used during the compliance test. For the purposes of these Application Notes, all references to the public IP addresses have been replaced with private addresses.

1.1. Phone Number Assignments For Testing

Typically, an enterprise has a set of consecutive DID numbers assigned to it from the local PSTN service provider. These same numbers would be provided to Tello Connect when subscribing to the service, so Tello Connect can route calls to these numbers. Tello Connect requires the use of E.164 formatted numbers in the SIP signaling messages. In the United States, these numbers are of the form 1 + area code + 7 digits.

For the compliance test, however, only one real PSTN routable number was available to assign to each site. These numbers were in the 732 area code. In order to test additional numbers, numbers in the 700 area code were arbitrarily assigned to each site. Numbers in the 700 area code are currently not used by the PSTN, so these numbers were guaranteed not to collide with numbers of real Tello Connect customers as they passed through the service. In addition, they aided the testing by the fact that if the call completed, the call had to have been completed by the Tello Connect service and not the PSTN. Numbers of the form 1-700-557-xxxx were assigned to Subscriber A. Numbers of the form 1-700-556-xxxx were assigned to Subscriber B.

These Application Notes do not show all calls being routed to Tello Connect as would be done in a typical customer environment. Due to a restriction in the test environment unrelated to this solution, a more limited subset of numbers was routed to Tello Connect. This subset included the numbers assigned to each site plus numbers in the 732, 303, and 800 area codes. This range of numbers allowed calls to be made to local, long distance, and toll-free non-subscribers as well as the two subscriber sites. In addition, for purposes of security, specific configuration screens involving the public PSTN numbers are not shown. However, these screens are described using the numbers in the 700 area code as examples.

1.2. SIP Call Flow

To better understand how calls are routed between the two enterprise sites shown in **Figure 1**, a typical call flow is described in this section. **Figure 2** shows the logical call flow of a call originating from an Avaya SIP Telephone at Subscriber A and terminating on an Avaya SIP Telephone at Subscriber B.

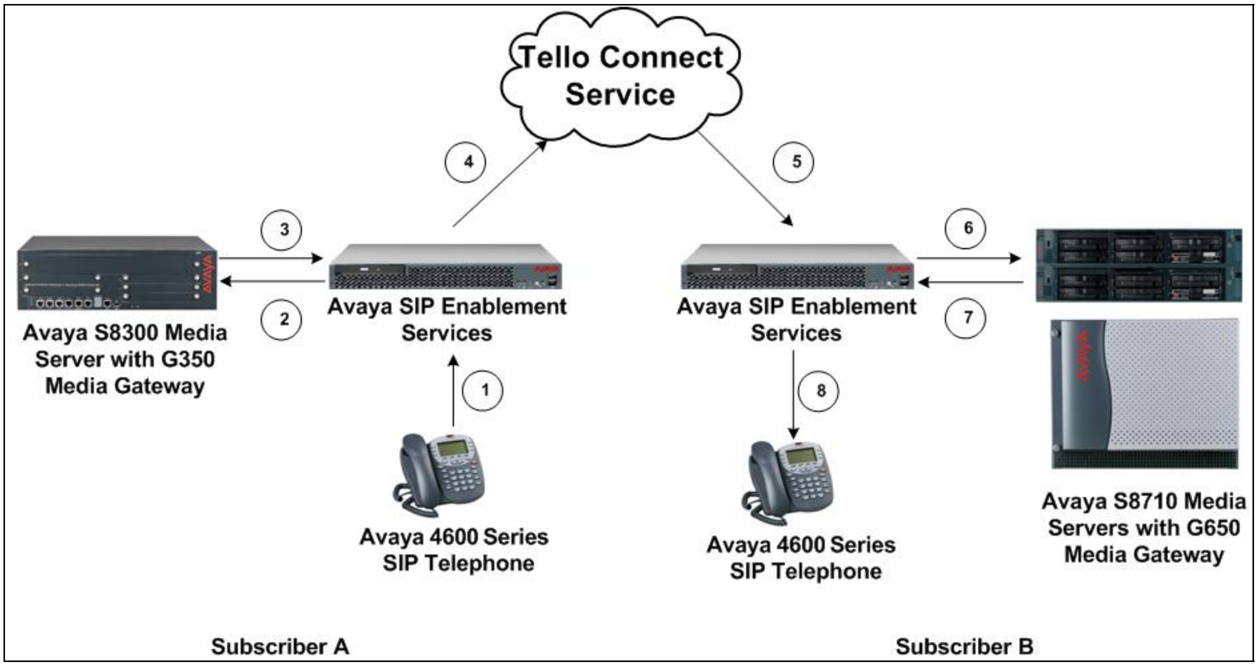

Figure 2: SIP Call Flow Between Two Tello Connect Subscribers

The details of the call flow are described below:

1. An Avaya SIP telephone at Subscriber A originates a call to a user at Subscriber B. The call request is delivered to Avaya SES. If the originator were an H.323, digital or analog endpoint, the call request would be sent to Avaya SES from Avaya Communication Manager.
2. Avaya SES routes the call over a SIP trunk to the Avaya S8300 Media Server running Avaya Communication Manager for origination services. This allows Avaya Communication Manager to apply the appropriate call restrictions to the endpoint, handle call routing, and track the status of the SIP telephone, which is an off-PBX station.
3. After applying the origination services, Avaya Communication Manager routes the call back to Avaya SES in the same manner.
4. Avaya SES routes the call to Tello Connect via SIP trunking.
5. Tello Connect locates the destination phone number in the subscriber database and forwards the call to Avaya SES at Subscriber B also via SIP trunking.
6. Avaya SES routes the call to the Avaya S8710 Media Server running Avaya Communication Manager over a SIP trunk.

7. Since the call is destined for an Avaya SIP telephone, Avaya Communication Manager routes the call back to Avaya SES in the same manner. If the destination of the call was an H.323, digital or analog endpoint, Avaya Communication Manager would terminate the call directly to an endpoint and steps 4 and 5 would not be required.
8. Avaya SES terminates the call to the Avaya SIP telephone.

2. Equipment and Software Validated

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

Equipment/Software	Version
Avaya S8300 Media Server with Intuity Audix 770 (IA770)	Communication Manager 3.0.1 (R013x.00.1.346.0)
Avaya G350 Media Gateway (Media Gateway Processor)	24.21.1
Avaya S8710 Media Servers	Communication Manager 3.0 (R013x.00.0.340.3) with the following updates: 340.3 – 10101 340.3 – 10673
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface (IPSI)	HW 03 FW 22
TN799DP C-LAN Interface (C-LAN)	HW 01 FW 16
TN2302AP IP Media Processor (MEDPRO)	HW 20 FW 107
Avaya SIP Enablement Services (SES)	3.0 (3.0.0.0-31.0) same at each site
Avaya 4600 Series SIP Telephones	2.2.2
Avaya 4600 Series H.323 IP Telephones	2.3
Avaya 6400D Series Digital Telephones	-
Avaya 6200 Series Analog Telephones	-
Tello Connect	1.5

3. Configure Avaya Communication Manager

This section describes the steps for configuring Avaya Communication Manager to interoperate with Tello Connect. It is important to reiterate that there is no direct SIP signaling path between Tello Connect and Avaya Communication Manager or the SIP endpoints. All SIP messages flowing to and from Tello Connect are directed to Avaya SES which acts as a SIP proxy for Avaya Communication Manager and the SIP endpoints. As a result, the configuration of Avaya Communication Manager is focused on two particular areas.

The first area is the establishment of a SIP trunk group between Avaya Communication Manager and Avaya SES. This trunk group will carry the SIP signaling and RTP voice packets between Avaya Communication Manager and Avaya SES including packets bound to and from Tello Connect.

The second area is the configuration of SIP endpoints as Outboard Proxy SIP (OPS)[1] stations. OPS 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. A high-level view of this call flow was previously shown in **Figure 2.** As a result, SIP signaling traffic to and from the SIP endpoints also flow across the SIP trunk group between Avaya Communication Manager and Avaya SES. It is not necessary to have SIP endpoints in order to use SIP trunking to Tello Connect. The steps discussed in Sections 3.11 and 3.12 describing SIP endpoint administration may be omitted if SIP endpoints are not used.

The following configuration of Avaya Communication Manager was performed using the System Access Terminal (SAT). The configuration was performed at each subscriber site in the test configuration. After completion of the configuration in this section, perform a **save translations** command to make the changes permanent.

3.1. Verify System Capacities

Using the SAT, verify there exists sufficient SIP Trunks and Off-PBX Telephones capacities by displaying the **System-Parameters Customer-Options** form shown in **Figure 3**. 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 make the appropriate changes.

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 used.

[1] Depending on the Avaya server product, the acronym OPS stands for two different feature names that are functionally equivalent. For Avaya SIP Enablement Services, the extended features capability is referred to as Outboard Proxy SIP. This capability is provided by Avaya Communication Manager as part of a more general feature extension package known as Off-PBX Stations, which can be applied to other remote devices such as cell phones. For that reason, the administration screens in this section will refer to the latter name or "off-pbx-telephone". For the purposes of the Avaya SIP offer and these Application Notes, the terms can be used interchangeably.

```
display system-parameters customer-options                      Page   1 of  10
                                OPTIONAL FEATURES
     G3 Version: V13
       Location: 1                             RFA System ID (SID): 1
       Platform: 13                            RFA Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 900   66
                                      Maximum Stations: 40    24
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 50    0
                    Maximum Off-PBX Telephones -   OPS: 50    10
                    Maximum Off-PBX Telephones - SCCAN: 0     0
        (NOTE: You must logoff & login to effect the permission changes.)
```

Figure 3: 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 number of SIP trunks used. Each SIP call between two SIP destinations (whether internal or external) requires two SIP trunks for the duration of the call. Thus, a call from a SIP telephone to a SIP service provider will use two SIP trunks, as will a call between two local SIP extensions. A call between a non-SIP telephone and a SIP service provider will only use one trunk.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 100   12
           Maximum Concurrently Registered IP Stations: 40    1
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 100   24

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

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

Figure 4: System-Parameters Customer-Options Form – Page 2

3.2. Assign Node Names

In the **IP Node Names** form, assign the node name and IP address for Avaya SES at the enterprise site. In this case, *SES* and *10.1.1.86* are being used, respectively. The node name *SES* will be used throughout the other configuration forms of Avaya Communication Manager. In this example, *procr* and *10.1.1.84* are the name and IP address assigned to the Avaya S8300 Media Server.

```
change node-names ip                                            Page   1 of   1
                                  IP NODE NAMES
    Name              IP Address          Name              IP Address
SES                 10 .1  .1  .86                            .   .   .
default             0  .0  .0  .0                             .   .   .
msgserver           10 .1  .1  .73                            .   .   .
procr               10 .1  .1  .84                            .   .   .
                       .   .   .                              .   .   .
                       .   .   .                              .   .   .
( 4  of 4   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 5: IP Nodes Names Form

3.3. Define IP Network Region

The **IP Network Region** form defines the connectivity settings for all VoIP resources and IP endpoints within an IP region. Select an IP network region that will contain the Avaya SES server. The association between this IP network region and the Avaya SES server will be done on the **Signaling Group** form as shown in **Figure 8**. In the case of the compliance test, the same IP network region that contains the Avaya S8300 Media Server was selected to contain the Avaya SES server. By default, the Media Server is in IP Network Region 1.

In the **IP Network Region** form:

- The **Authoritative Domain** field is configured to match the domain name configured on Avaya SES. In this configuration, the domain name is *devcon.com*. This name will appear in the "From" header of SIP messages originating from this IP region.
- By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G350 Media Gateway. This is true for both intra-region and inter-region IP-IP Direct Audio. Shuffling can be further restricted at the trunk level on the **Signaling Group** form.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within this IP network region. In this configuration, this codec set will apply to calls using Tello Connect as well as any IP phone (H.323 or SIP) within the enterprise. If different IP network regions are used for the Avaya S8300 Media Server and Avaya SES, then Page 3 of each **IP Network Region** form must be used to specify the codec set for inter-region communications.
- The default values can be used for all other fields.

```
change ip-network-region 1                                          Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: devcon.com
    Name:
                                Intra-region IP-IP Direct Audio: yes
MEDIA PARAMETERS                Inter-region IP-IP Direct Audio: yes
      Codec Set: 3                         IP Audio Hairpinning? y
   UDP Port Min: 2048
   UDP Port Max: 3028                    RTCP Reporting Enabled? y
DIFFSERV/TOS PARAMETERS          RTCP MONITOR SERVER PARAMETERS
 Call Control PHB Value: 46       Use Default Server Parameters? y
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        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 6: IP Network Region Form

3.4. Define IP Codecs

Use the change **ip-codec-set *n*** command, where ***n*** is the codec set value specified in Section 3.3, to enter the supported audio codecs for calls routed to Avaya SES. Multiple codecs can be listed in priority order to allow the codec to be negotiated during call establishment. Since the media path does not pass through Tello Connect, Tello Connect does not place any requirements on which codecs can be used. The list should include the codecs the enterprise wishes to support within the normal trade-off of bandwidth versus voice quality. **Figure 7** shows the values used in the compliance test.

```
change ip-codec-set 3                                               Page   1 of   2

                          IP Codec Set

    Codec Set: 3

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

Figure 7: IP Codec Set Form

3.5. Configure the Signaling Group

Configure the **Signaling Group** form shown in **Figure 8** 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 communication between Avaya SES and Avaya Communication Manager.
- In the case of Subscriber A in **Figure 1**, specify the Avaya S8300 Media Server (node name *procr*) and the Avaya SES server (node name *SES*) as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** fields, respectively. These field values are taken from the **IP Node Names** form shown in **Figure 5**. For other Media Server platforms such as the one used at Subscriber B, the near (local) end of the SIP signaling group will be a C-LAN board instead of the Media Server.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP network region value assigned in the **IP Network Region** form (**Figure 6**). This defines which IP network region contains the Avaya SES server. 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.
- Enter the domain name of Avaya SES 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" header in the INVITE message.
- The **Direct IP-IP Audio Connections** field must be set to *n* since this combined Avaya/Tello solution does not support shuffling of SIP calls. Disabling shuffling at the SIP trunk level while having it enabled at the IP network region level will allow non-SIP IP (H.323) calls to continue to be shuffled.
- The **DTMF over IP** field must be set to the default value of *rtp-payload* for a SIP trunk. This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833.
- The default values for the other fields may be used.

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

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


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

                                            Bypass If IP Threshold Exceeded? n

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

Figure 8: Signaling Group Form

3.6. Configure the Trunk Group

Configure the **Trunk Group** form as shown in **Figure 9** 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**) that is consistent with the dial plan.
- 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 8**.
- Specify the **Number of Members** supported by this SIP trunk group. As mentioned earlier, each SIP call between two SIP destinations (whether internal or external) requires two SIP trunks for the duration of the call. Thus, a call from a SIP telephone to a SIP service provider will use two SIP trunks, as will a call between two local SIP extensions. A call between a non-SIP telephone and a SIP service provider will only use one trunk.
- Use the default values for the other fields.

```
add trunk-group 1                                           Page   1 of  19
                                TRUNK GROUP

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

     Unicode Name? y

                                           Redirect On OPTIM Failure: 5000

            SCCAN? n                              Digital Loss Group: 18
```

Figure 9: 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.
- Use the default values for the other fields.

```
add trunk-group 1                                            Page   2 of  19
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y


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

                                                Replace Unavailable Numbers? n
```

Figure 10: Trunk Group Form – Page 2

3.7. Configure Calling Party Number Information

Use the **change public-unknown numbering** command to define the full calling party number to be sent to the far-end. Add an entry for the trunk group defined in Section 3.6. In the example shown below, all calls originating from a 5 digit extension beginning with 7 and routed across trunk group 1 will have the prefix 70055 prepended to the extension number to create a 10 digit calling number. For example, extension 70000 becomes 7005570000. This calling party number will be sent to the far-end in the SIP "From" header.

```
change public-unknown-numbering 0                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                   Total                                Total
Ext Ext     Trk       CPN           CPN Ext Ext    Trk       CPN          CPN
Len Code    Grp(s)    Prefix        Len Len Code   Grp(s)    Prefix       Len

 5  7       1         70055         10
```

Figure 11: Numbering Public/Unknown Format Form

3.8. Configure Incoming Digit Translation

Use the **change inc-call-handling-trmt trunk-group *n*** command, where n is the SIP trunk group number, to map incoming DID calls to the proper extension(s). For the compliance test, DID numbers of the form 1-700-557-xxxx were assigned to Subscriber A and numbers of the form 1-700-556-xxxx were assigned to Subscriber B. The example below shows the configuration at Subscriber A. The entry defines that all incoming calls on trunk group 1 with 11 digits starting with 170055 will have the first 6 digits deleted. The remaining 5 digits map directly to a local 5 digit extension on Avaya Communication Manager. For example, DID number 1-700-557-0001 maps to extension 70001.

```
change inc-call-handling-trmt trunk-group 1                     Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/       Called    Called        Del  Insert
 Feature         Len      Number
 tie              11    170055          6
```

Figure 12: Incoming Call Handling Treatment – Simple Extension Mapping

Each subscriber site also had a PSTN number in the 732 area code assigned to it. This number would also require manipulation of the incoming digits to map it to a specific extension. This is done by adding another entry to the **Incoming Call Handling Treatment** form shown in **Figure 12**. The specific mapping for this number is not shown.

3.9. Configure the Route Pattern

The compliance testing used Automatic Route Selection (ARS) to define route pattern 1 as the route for all outbound calls to Tello Connect. Furthermore, the test configured trunk group 4 as an ISDN-PRI trunk group to the PSTN. It is beyond the scope of these Application Notes to describe ARS or ISDN-PRI trunking. For more information on these topics see [1] and [2]. The purpose of this sub-section is to illustrate the route pattern configuration necessary to define the ISDN-PRI PSTN trunk as a secondary path for the SIP trunk servicing traffic to Tello Connect.

Figure 13 shows the route pattern used in the compliance test for all outbound traffic. There are two entries in the route pattern. The first is for trunk group 1 which is the SIP trunk group to Avaya SES defined in Section 3.6. The second is for trunk group 4 described above. The Look Ahead Routing (**LAR**) field must be set to *next* for the first entry, in order for Avaya Communication Manager to use the second entry in the table when Tello Connect returns a 406 error response. Tello Connect returns this response when the dialed number can not be located in the database.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: Tello
                             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: 4    0       1                                                    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                                        next
 2: y y y y y n  n            rest                                        none
 3: y y y y y n  n            rest                                        none
 4: y y y y y n  n            rest                                        none
 5: y y y y y n  n            rest                                        none
 6: y y y y y n  n            rest                                        none
```

Figure 13: Route Pattern

3.10. Assign Route Pattern to Location

Use the **change locations** command to assign the route pattern to the location. Only one location created by default, known as Main, exists for each subscriber site. Enter the route pattern number from the previous section in the **Proxy Sel. Rte. Pat.** field. Use the default values for all other fields.

```
change locations                                              Page   1 of   4
                                   LOCATIONS

                ARS Prefix 1 Required For 10-Digit NANP Calls? y

 Loc.  Name           Timezone Rule  NPA  ARS   Attd         Pre-   Proxy Sel.
 No.                   Offset             FAC   FAC          fix    Rte. Pat.
 1:   Main            + 00:00   0                                   1
 2:
 3:
```

Figure 14: Locations

3.11. Assign a Station

The procedure in this sub-section is only necessary if the configuration uses SIP endpoints as OPS stations. Repeat the procedure for each SIP endpoint in the configuration.

Create an OPS station by using the **add station *x*** command where ***x*** is the extension to be assigned to the station. Enter the field values as described below.

- Leave the station **Type** at the default *6408D+* value. This is the Avaya recommended best practice that will prevent an alarm warning that occurs when 4600 series telephone models are entered.
- Enter *X* in the **Port** field to indicate that no Avaya Communication Manager port hardware is associated with this station.
- 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. The Class of Restrictions (**COR**) and Class of Service (**COS**) will govern the features and call restrictions that apply to this station. This station is also configured with voicemail coverage via coverage path 1 using the Intuity Audix 770, an option integrated into the Avaya S8300 Media Server. For additional information on configuring Intuity Audix 770, refer to [4].

```
add station 70000                                             Page   1 of   4
                                     STATION

Extension: 70000                          Lock Messages? n          BCC: 0
     Type: 6408D+                         Security Code:             TN: 1
     Port: X                            Coverage Path 1: 1          COR: 1
     Name: SIP70000                     Coverage Path 2:            COS: 1
                                        Hunt-to Station:
STATION OPTIONS
              Loss Group: 2             Personalized Ringing Pattern: 1
             Data Module? n                         Message Lamp Ext: 70000
            Speakerphone: 2-way                  Mute Button Enabled? y
        Display Language: English
                                                   Media Complex Ext:
                                                        IP SoftPhone? n
```

Figure 15: Station Administration – Page 1

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

- By default, the last call appearance is reserved for outgoing calls from the phone. If it is desirable to allow an incoming call to use the last available call appearance when all others are occupied, set the **Restrict Last Appearance** value to *n*. In this mode, all call appearances are available for making or receiving calls.
- Verify that the **Per Station CPN – Send Calling Number** field is set to *y* or blank to allow calling party number information to be sent to the far-end when placing outgoing calls from this station. The default value for this field is blank.

```
add station 70000                                             Page   2 of   4
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe             Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
             CDR Privacy? n                              Data Restriction? n
   Redirect Notification? y                    Idle Appearance Preference? n
 Per Button Ring Control? n                  Bridged Idle Line Preference? n
   Bridged Call Alerting? n                      Restrict Last Appearance? n
  Active Station Ringing: single         Conf/Trans on Primary Appearance? n

        H.320 Conversion? n         Per Station CPN - Send Calling Number? y
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
    MWI Served User Type: qsig-mwi             Display Client Redirection? n
                                              Select Last Used Appearance? n
                                                Coverage After Forwarding? s

 Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
 Emergency Location Ext: 70000     Always Use? n      IP Audio Hairpinning? y
```

Figure 16: Station Administration – Page 2

On Page 3 of the **Station** form, configure the appropriate number of call appearances for the SIP telephone as shown in **Figure 17**. To ensure proper transfer and conference operation, the number of *call-appr* on the **Station** form must be at least 3 and must be one more than the number of call appearances defined on the SIP Telephone in the *46xxxsettings.txt* configuration file. The default value in the configuration file is 3 so the station form typically contains 4. Discussion of settings in the configuration file is beyond the scope of these Application Notes. Additional information is available in [6].

```
add station 70000                                              Page   3 of   4
                                     STATION
 SITE DATA
       Room:                                              Headset? n
       Jack:                                              Speaker? n
      Cable:                                             Mounting: d
      Floor:                                         Cord Length: 0
   Building:                                           Set Color:

ABBREVIATED DIALING
     List1:                    List2:                    List3:

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

Figure 17: Station Adminstration – Page 3

3.12. Configure Off-PBX Station Mapping

The procedure in this sub-section is only necessary if the configuration uses SIP endpoints as OPS stations. Repeat the procedure for each SIP endpoint in the configuration.

Configure the **Off-PBX Telephone Station-Mapping** form so that calls destined for a SIP telephone at the enterprise site are routed to Avaya SES, which will then forward the call to the SIP telephone.

On the **Off-PBX-Telephone Station-Mapping** form shown in **Figure 18**:

- Specify the **Station Extension** of the SIP endpoint.
- Set the **Application** field to *OPS*.
- Set the **Phone Number** field to the digits to be sent over the SIP trunk. In this case, the SIP telephone extensions configured on Avaya SES also match the extensions of the corresponding OPS stations on Avaya Communication Manager. However, this is not a requirement.
- Set the **Trunk Selection** field 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 9**.
- Set the **Configuration Set** to the value ‘1’, which in these Application Notes uses the default values of the Configuration Set form.

```
change off-pbx-telephone station-mapping 70000                    Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station     Application   Dial   Phone Number      Trunk       Configuration
 Extension                 Prefix                   Selection   Set
 70000         OPS              - 70000              1          1
                                -
```

Figure 18: 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 *4*, which corresponds to the number of call appearances configured on the station form. Accept the default values for the other fields.

```
change off-pbx-telephone station-mapping 70000                  Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged
 Extension     Limit      Mode         Allowed     Calls
 70000         4          both         all         both
```

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

4. Configure Avaya SES

This section covers the administration of Avaya SES. Avaya SES is configured via an Internet browser using the administration web interface. It is assumed that Avaya SES software and the license file have already been installed on the server. During the software installation, the installation 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 [3].

The configuration of Avaya SES is divided into two parts: Section 4.1 - 4.9 provides the steps necessary to configure SIP trunking to Tello Connect. Section 4.10 provides the steps necessary to administer optional SIP endpoints. This configuration was performed on both enterprise sites in the test configuration.

4.1. Log in to Avaya SES

Access the Avaya 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 SES.

Log in with the appropriate credentials and then select the **Launch Administration Web Interface** link from the main page as shown in **Figure 20**.

Figure 20 - Avaya SES Main Page

The Avaya SES Administration Home Page shown in **Figure 21** will be displayed.

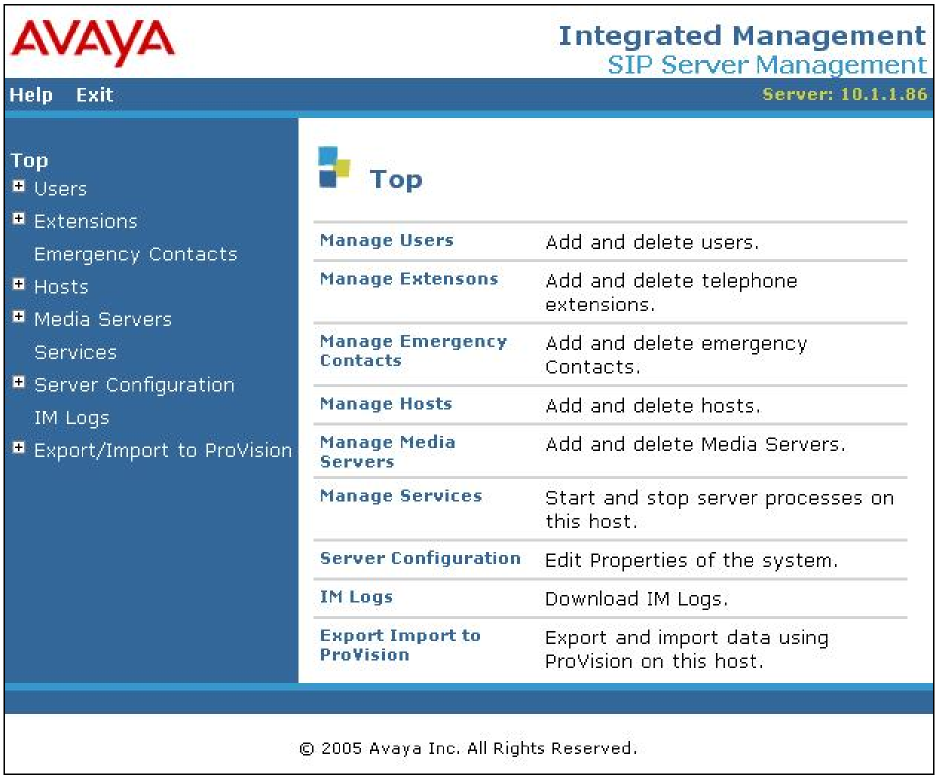

Figure 21: Avaya SES Administration Home Page

4.2. Saving 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 Avaya SES administration pages as shown in **Figure 22**. It is recommended that this be done after making each set of changes described in the following sub-sections.

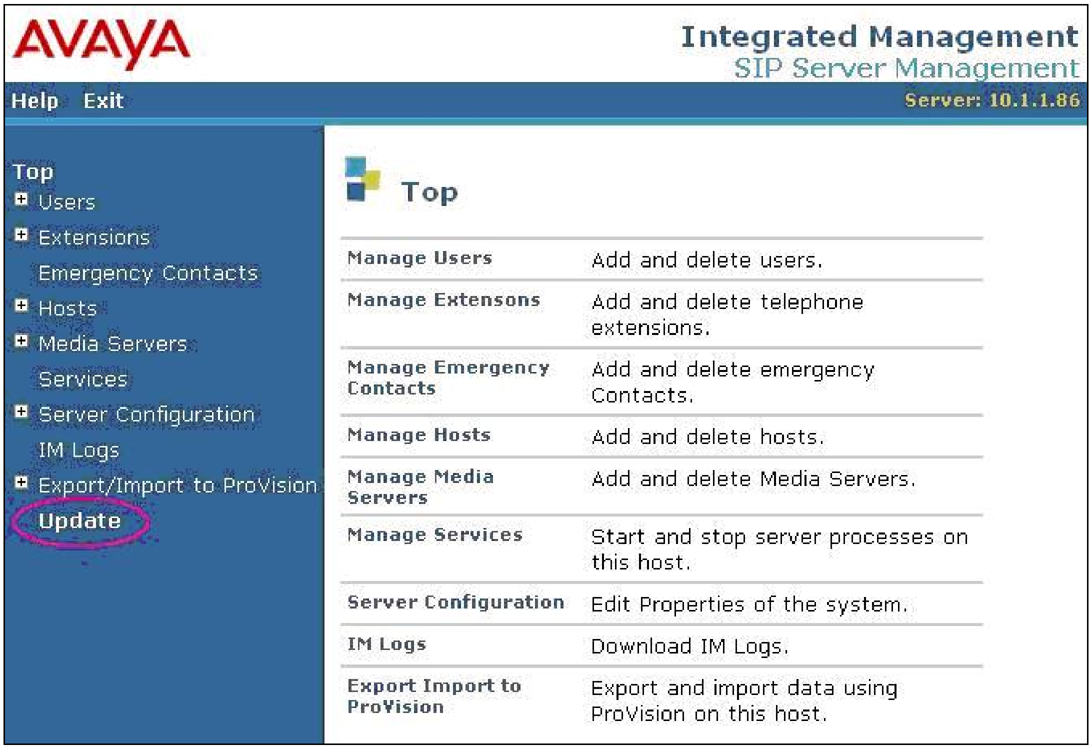

Figure 22: Update Avaya SES Administrative Changes

4.3. Define System Properties

From the left pane of the administration web interface, expand the **Server Configuration** option and select **System Properties**. This page displays the software version in the **CCS version** field and the network properties entered via the installation script during the installation process.

On the **Edit System Properties** page:

- Enter the **SIP Domain** name assigned to Avaya SES. This must match the Authoritative Domain field configured on Avaya Communication Manager shown in **Figure 6**.
- Enter the **License Host** field. This 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 **Edit System Properties** page, click the **Update** button.

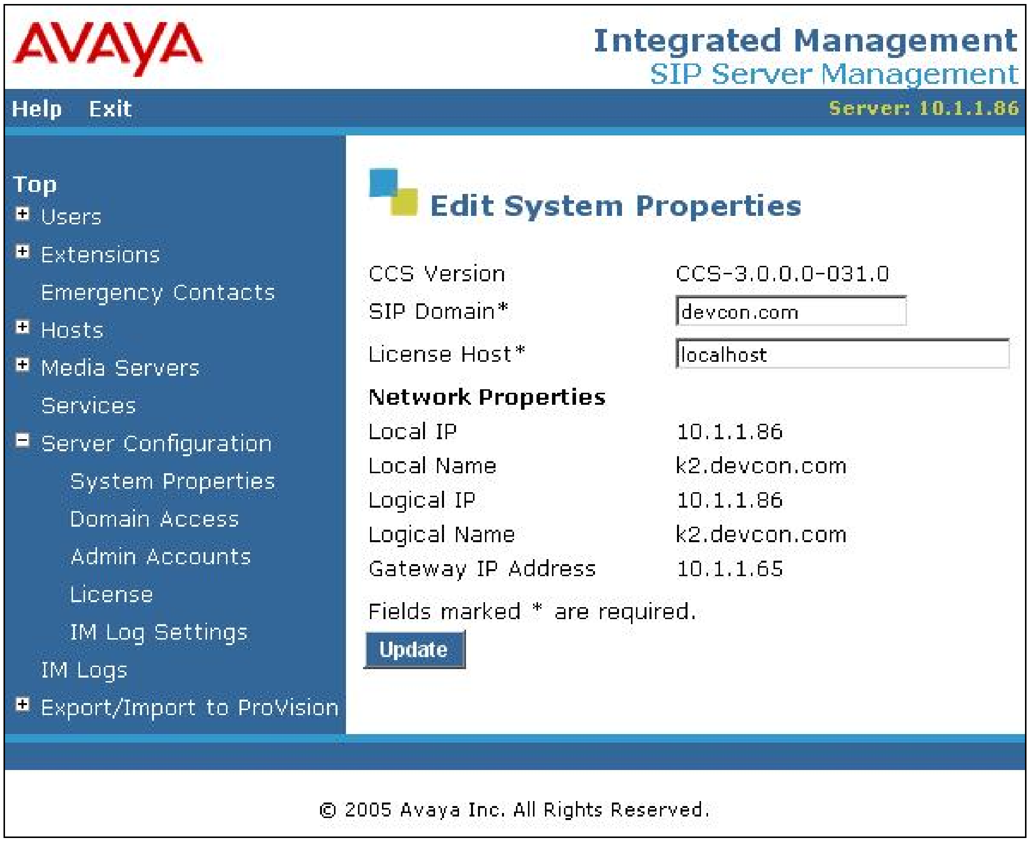

Figure 23: Edit System Properties

4.4. Enter Avaya SES Host Information

After setting up the domain on the **Edit System Properties** page, create a host computer entry for Avaya SES. The following example shows the **Edit Host** page since the host had already been added to the system.

The **Edit Host** page 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. The **Logical Name** is the fully qualified domain name.
- Enter the **DB Password** that was specified while running the installation script during the system installation.
- Configure the **Host Type** field. Since only one Avaya SES proxy server exists in the enterprise network of the test configuration, the Avaya SES server provides the functionality of both a *home* and *edge* server. Thus, the **Host Type** is configured as *home/edge*.
- The default values for the other fields may be used as shown in **Figure 24**.
- Click the **Update** button.

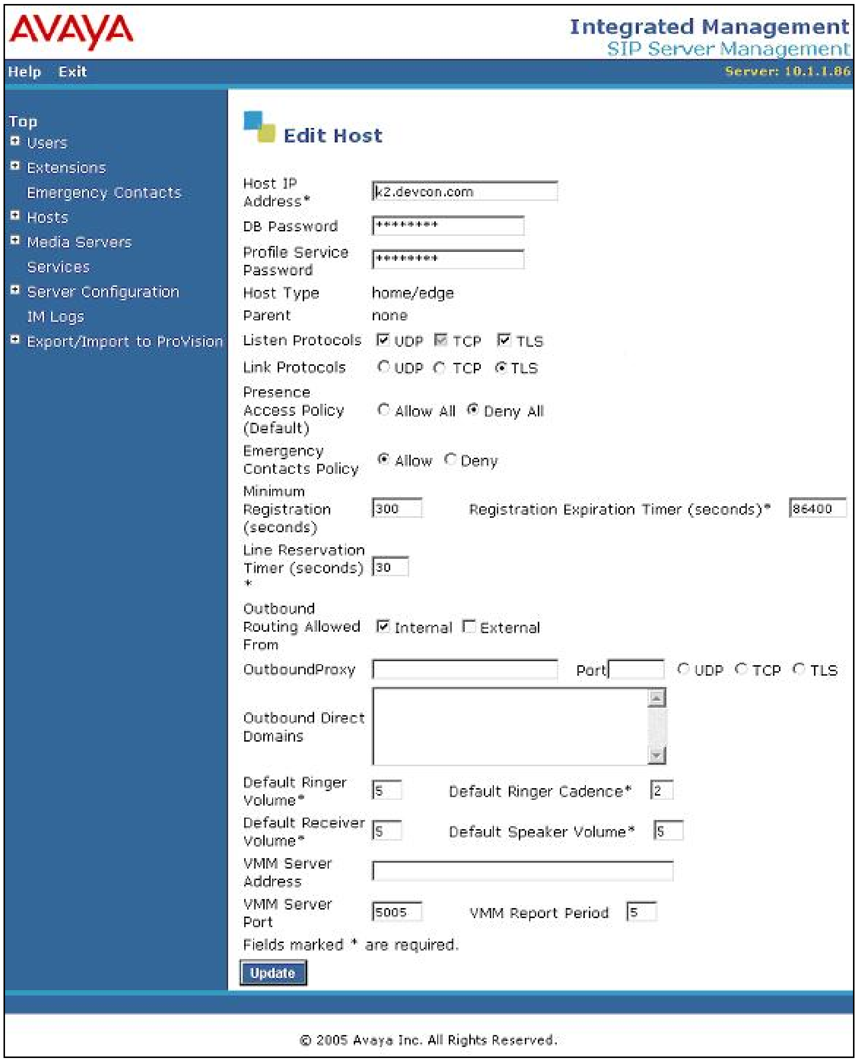

Figure 24: Edit Host

4.5. Add a Media Server

From the left pane of the administration web interface, expand the **Media Server** option and select **Add** to add the Avaya Media Server to the list of media servers known to Avaya SES. Adding the media server will create the Avaya SES side of the SIP trunk previously created in Avaya Communication Manager.

On the **Add Media Server** page, enter the following information:

- A descriptive name in the **Media Server Interface** field (i.e. S8300).
- In the **Host** field, select the name of the Avaya SES server from the pull-down menu that will serve as the SIP proxy for this media server. Since there is only one Avaya SES server in this configuration, the **Host** field is set to the host name shown 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 communication between Avaya SES and Avaya Communication Manager.

- Enter the IP address of the Avaya S8300 Media Server in the **SIP Trunk IP Address** field. In other media server platforms, such as an Avaya S8710 Media Server using an Avaya G650 Media Gateway, the **SIP Trunk IP Address** would be the IP address of the C-LAN board.
- Use the default values for all other fields.
- After completing the **Add Media Server** page, click on the **Add** button.

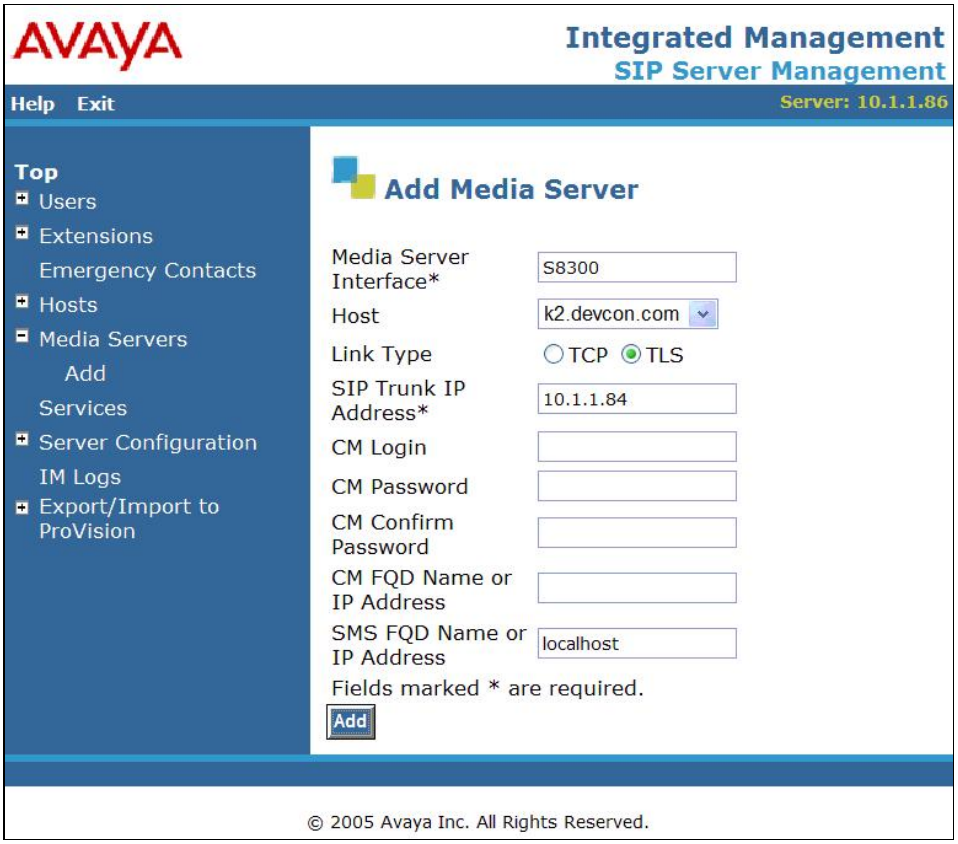

Figure 25: Add Media Server

4.6. Specify Address Maps to Media Servers

Inbound SIP calls arriving at Avaya SES are routed to the appropriate Avaya Communication Manager for termination services. This routing is specified in a Media Server Address Map configured on Avaya SES.

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 Media Servers running Avaya Communication Manager supported by Avaya SES.

In this test configuration, only incoming calls from Tello Connect require a Media Server

Address Map entry. Calls originated by Avaya SIP telephones configured as OPS stations are automatically routed to the proper Avaya Communication Manager by the assignment of a 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 Tello Connect, the *user* portion of the SIP URI will contain the incoming direct inward dialed telephone number.

An example of a SIP URI in an INVITE message received from Tello Connect would be:

sip: 17005570001@10.1.1.86.

In this example, the user portion is the called party number 17005570001.

For the compliance test, phone numbers beginning with the prefix of 1700557 were assigned to the enterprise site of Subscriber A. Phone numbers beginning with the prefix of 1700556 were assigned to Subscriber B.

Thus, the media server address map strategy was to define pattern matches for the 7-digit prefix in the URI and have Avaya SES forward the messages that match to the appropriate 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** page.
- Click on the **Map** link associated with the appropriate media server to display the **List Media Server Address Map** page.
- Click on the **Add Map In New Group** link. The page shown in **Figure 26** is displayed. The **Host** field displays the name of the media server to which this map applies.
- Enter a descriptive name in the **Name** field.
- Enter the regular expression to be used for the pattern matching in the **Pattern** field. The example in **Figure 26** shows the pattern specification for DID numbers assigned to Subscriber A: *`^sip:1700557`*. This expression will match any SIP URI that begins with the text string *`sip:1700557`*. Based on the value of the **Host** field, these SIP calls will then be routed to host S8300. Appendix A provides a detailed description of the syntax for address map patterns.
- Click the **Add** button once the form is completed.

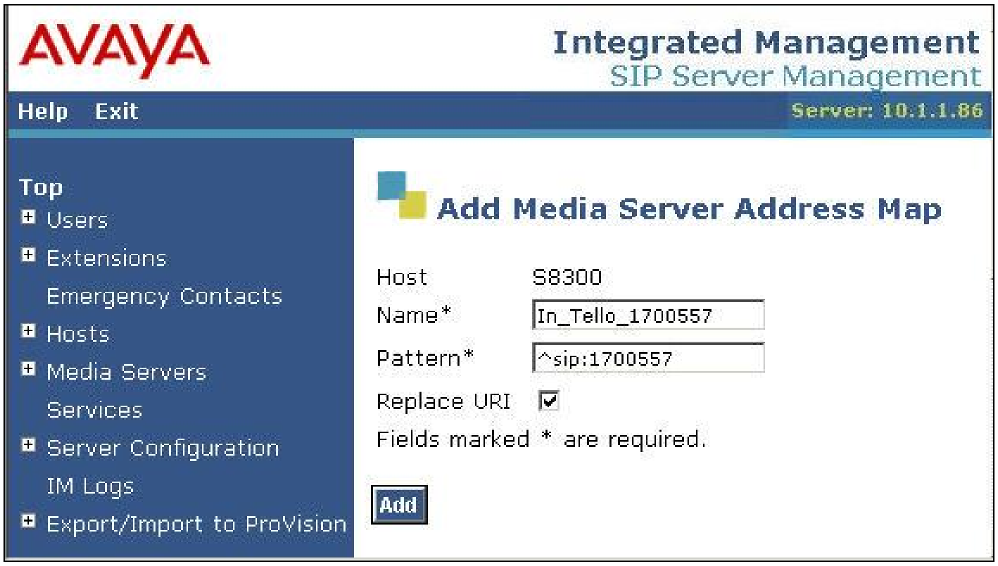

Figure 26: Media Server Address Map

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

Figure 27: List Media Server Address Map

Note that after the first Media Server Address Map is added, the Media Server Contact is created automatically. For the Media Server Address Map added in **Figure 26**, the following contact was created and displayed in the **Contact** field:

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

The contact specifies the IP address of the Avaya S8300 Media Server and the transport protocol used to send SIP signaling messages. The user in the original request URI is substituted for `$(user)`.

4.7. Specify Address Maps to Tello Connect

Outbound SIP calls are first directed by Avaya Communication Manager routing decisions to the SIP trunk group. These calls are then subject to further routing decisions determined by the Host Address Maps in Avaya SES. Similar to the inbound Media Server Address Maps, these Host Address Maps use pattern matching to direct outbound SIP messages to the proper destination. Furthermore, to ensure correct routing of calls by Avaya SES, the Host Address Maps and Media Server Address Maps must be mutually exclusive. Stated differently, any sequence of dialed digits or received digits in an external SIP call should match only one address map.

In the general case, the Host Address Map routing rule would be defined to send all outbound traffic to Tello Connect. To keep this rule from overlapping the pattern matched by the Media Server Address Map, it would also have to exclude the numbers which are matched by the pattern in **Figure 26**.

However, for the compliance test, a more limited set of dialed digits were routed to Tello Connect. They included the set of numbers that start with 170055 and numbers with area codes 732, 303, and 800. The example below shows the Host Address Map on Subscriber A that routes numbers beginning with 170055 but exclude the range defined in the Media Server Address Map in **Figure 26**. Additional Host Address Maps were added to cover the other dialing patterns described above.

It should be noted that a user dialed access code such as a 9 to place an outbound call is deleted by Avaya Communication Manager prior to routing the call to Avaya SES. Thus, these access codes do not appear in the matching patterns.

In addition, other Host Address Maps could be added if necessary to selectively route SIP traffic to different destinations such as different SIP proxies serving different geographic regions.

The configuration of the Host Address Map for all calls starting with digits 170055 and followed by any digit but 7 is shown in **Figure 28**.

- Access the **Add Host Address Map** page 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** page is displayed.
- From this page, click the **Add Map In New Group** link to display the **Add Host Address Map** page shown in **Figure 28**.
- Enter a descriptive name for the map, such as *Out_Tello_170055*.
- Specify an appropriate pattern for the call type. In this example, the pattern is `^sip:170055[0-6,8-9]`
- Leave the **Replace URI** checkbox selected.
- Click the **Add** button.

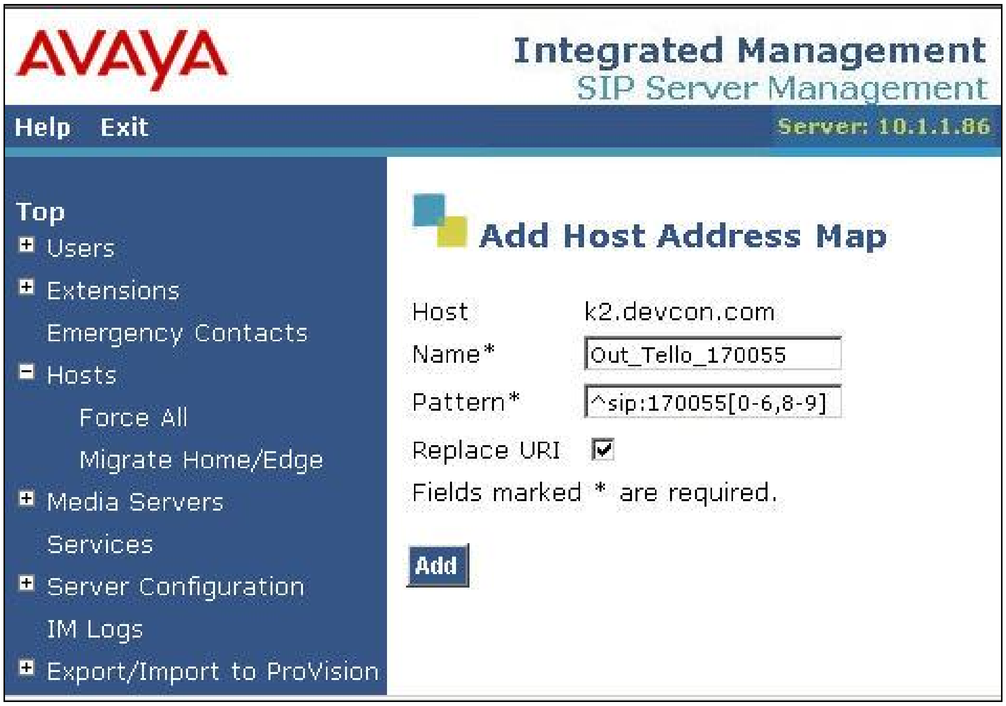

Figure 28: Edit Host Map Entry

Additional Host Address Map patterns are added in the same manner. The compliance test used various additional mappings for specific test cases including the following:

Name	Description
Out_Tello_1732	Maps all numbers with 732 area code to Tello Connect but excludes the 732 number assigned to this subscriber.
Tello_Tollfree	Maps all 800 area codes to Tello Connect - ^sip:1800
Tello_AC_303	Maps all 303 area codes to Tello Connect - ^sip:1303

Figure 29: Multiple Address Maps

4.8. Specify the Tello Connect Host Contact Information

The IP address for the Tello Connect SIP proxy must be administered in Avaya SES. In the example shown below, the IP address 20.1.1.54 is used. The actual public IP address of the Tello Connect proxy will be provided to the service subscriber by Tello.

To enter the SIP proxy information:

- As described in Section 4.7, display the **List Host Address Map** page.
- 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** page. On this page, the **Contact** field specifies the destination for the call and it is entered as:

```
sip:$(user)@20.1.1.54:5060;transport=udp
```

The user part in the original request URI is inserted in place of the "$(user)" string before the message is sent to Tello Connect.

- Click the **Add** button when completed.

After configuring the host contact information, the **List Host Address Map** page will appear as shown in **Figure 30**.

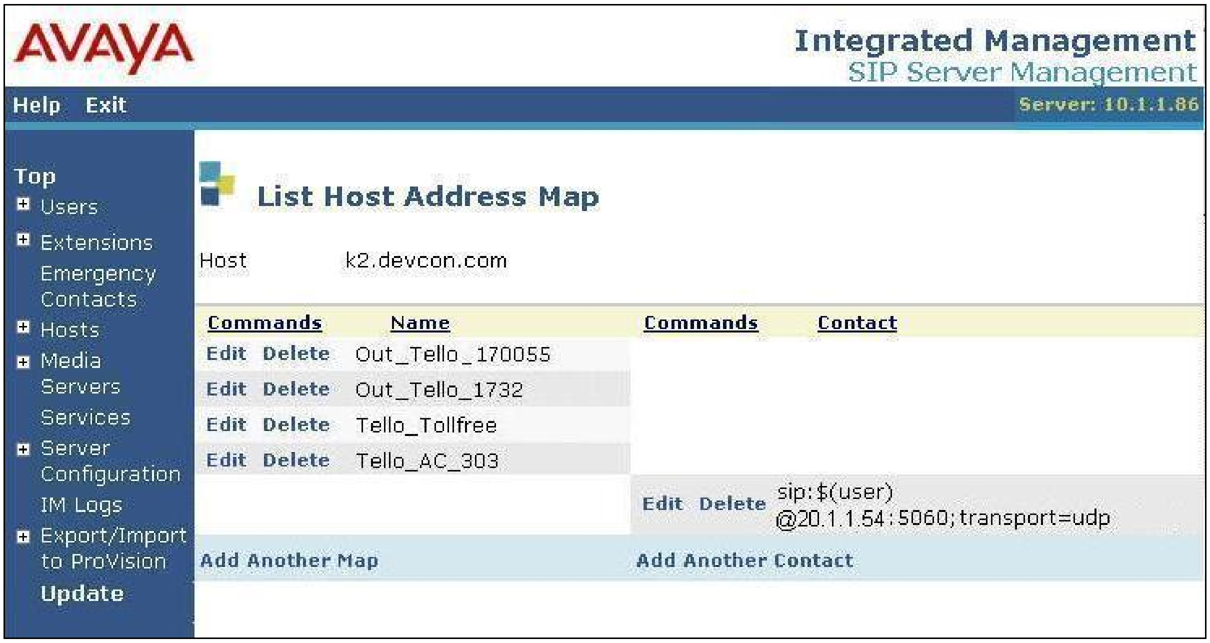

Figure 30: List Host Address Map

4.9. Specify the Tello Connect SIP Proxy as a Trusted Host

Complete the administration of Avaya SES by designating the IP address of the Tello Connect SIP Proxy as a trusted host. As a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the designated IP address.[2]

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:

- Telnet to the Avaya SES IP address (10.1.1.86) and log in using the administrative login and password.

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

- Enter the following **trustedhost** command at the Linux shell prompt.

  `trustedhost –a 20.1.1.54 –n k2.devcon.com –c TelloProxy1`

  The –a argument specifies the address to be trusted; –n specifies the Avaya SES host name; –c adds a comment.
- Use the following **trustedhost** command to verify the entry is correct.

  `trustedhost –L`

  Figure 31 illustrates the results of the **trustedhost** commands.[3]
- **Important Note:** Complete the trusted host configuration by returning to the main Avaya SES administration web interface and clicking on the **Update** link as shown in **Figure 22**. If the **Update** link is not visible, refresh the page by selecting **Top** from the left hand menu. This step is required even though the trusted host was configured via the Linux shell.

```
admin@k2> trustedhost -a 20.1.1.54 -n k2.devcon.com –c TelloProxy1
20.1.1.54 is added to trusted host list.

admin@k2> trustedhost -L
Third party trusted hosts.
       Trusted Host       |       CCS Host Name       |         Comment
--------------------------+---------------------------+-------------------------
20.1.1.54                 | k2.devcon.com             |    TelloProxy1
```

Figure 31: Configuring a Trusted Host

4.10. Add a SIP User With Avaya Communication Manager Extension

The procedure in this sub-section is only necessary if the configuration uses SIP endpoints. Repeat the procedure for each SIP endpoint in the configuration.

Create the SIP user record as follows:

- In the Avaya SES administration web interface, expand the **Users** link in the left side blue navigation bar and click on the **Add** link.
- On the **Add User** page, 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 SES server hosting the domain (*devcon.com*) for this user.
- Enter the **First Name** and **Last Name** of the user.

---

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

`trustedhost –d 20.1.1.54 –n k2.devcon.com`

removes the trust relationship added above.

- 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.
- Use default values for all other fields.
- Press the **Add** button. This will cause a confirmation page to appear.
- Press **Continue** on the confirmation page.

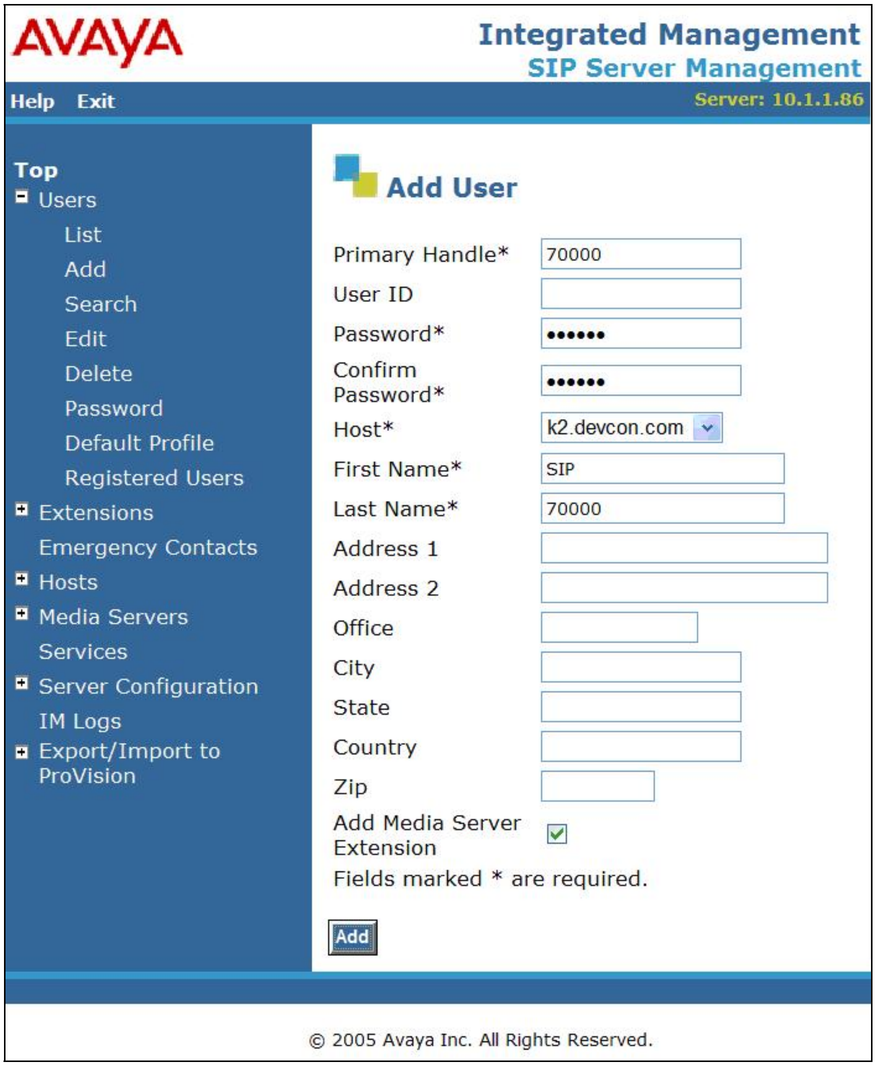

Figure 32: Add User

The **Add Media Server Extension** page will appear as shown in **Figure 33**. This page is used to associate the SIP phone handle to the corresponding extension on Avaya Communication Manager.

- On the **Add Media Server Extension** page, enter the **Extension** configured on the media server, shown in **Figure 15**, for the OPS extension on Avaya Communication Manager. It is recommended that the media server extension be the same as the user extension but it is not required.
- Select the **Media Server** assigned to this extension.

- Click on the **Add** button.
- To commit the configuration changes, click on the **Update** link in the left pane.

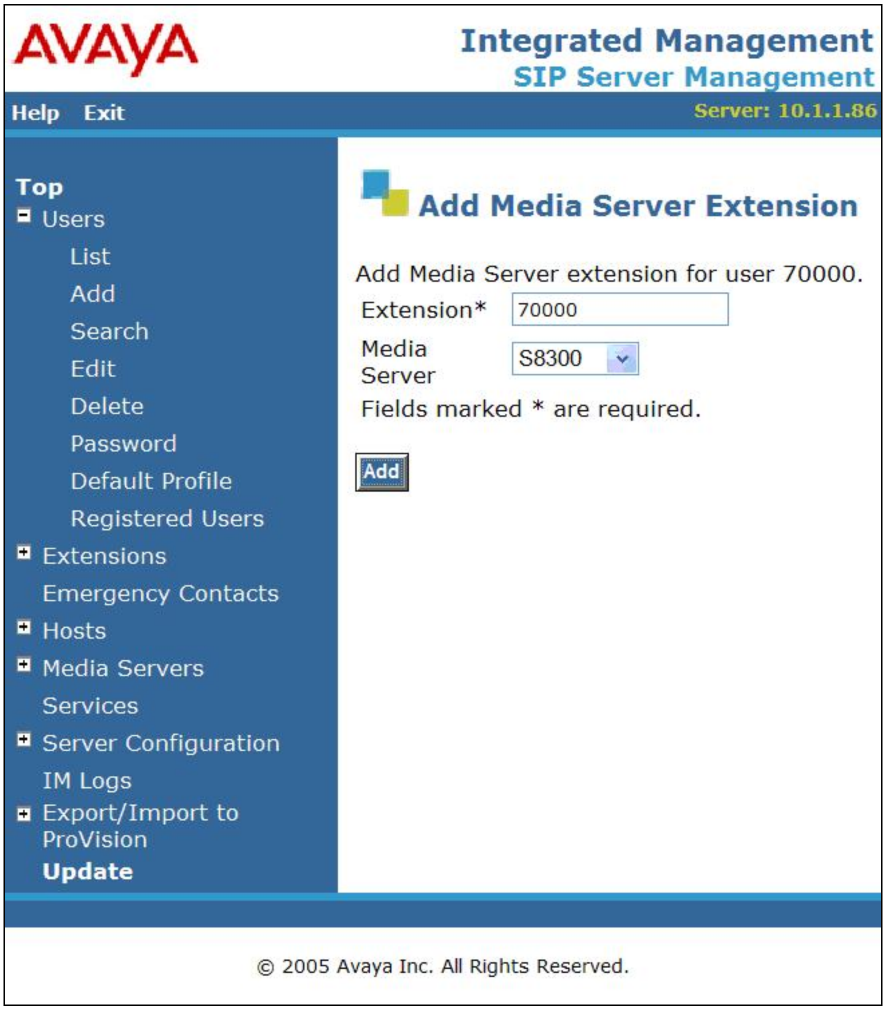

Figure 33: Add Media Server Extension

5. Configure Tello Connect Service

In order to use the Tello Connect service, a customer must subscribe to the service using the Tello Connect sales process. For information on subscribing to Tello Connect, visit the sales links on the corporate web site at http://www.tello.com.

During the subscription process, certain information must be provided by the subscriber to Tello and vice versa. This information is summarized in the table below. All configuration of the service will be performed by Tello. A username and password will be provided by Tello for accessing the Tello Connect subscriber web site. This can be used by the subscriber to view the configuration information for his/her account and to make changes if necessary.

The values shown in the table are those used during the interoperability compliance test. Parameters related to the media stream, such as codec values and the use of RFC2833 for passing DTMF tones, do not have to be coordinated with Tello since the Tello Connect service does not touch the media stream. The media stream parameters are defined by the capabilities of the endpoints at each end of the call.

Data Required	Configuration
Listed Directory Numbers Owned by the Subscriber	Listed directory numbers owned by the subscriber need to be provided to Tello. This allows Tello to direct incoming calls to these numbers to the subscriber. In this configuration, listed directory numbers beginning with prefix 1-700-557 were assigned to endpoints at Subscriber A. Listed directory numbers beginning with prefix 1-700-556 were assigned to endpoints at Subscriber B. In addition, a real PSTN routable number in the 732 area code was also assigned to each site.
Subscriber Provides IP Address of Avaya SES	The IP address of Avaya SES in the enterprise network needs to be a public IP address. For purposes of these Application Notes, this address has been represented by the IP address 10.1.1.86. Tello uses this IP address for routing calls destined to the listed directory numbers assigned to the enterprise site.
Tello Provides Proxy IP Address	The IP address of the Tello SIP proxy is a public IP address that is provided by Tello to the subscriber. For the purposes of these Application Notes, this address has been represented by the IP address 20.1.1.54 and used to configure the host address maps in Avaya SES.
SIP Transport Protocol and Port	SIP signaling was transported between Avaya SES and Tello Connect using UDP and port 5060.

Data Required	Configuration
Username and Password for Subscriber Web Site	A username and password is provided by Tello for logging into the subscriber web site.

5.1. Log into the Subscriber Web Site

To view or change information relating to the subscriber account or configuration, access the subscriber web site at the URL provided by Tello during the subscription process. The main login page will appear. Log in to the site with a valid username and password. Click **Submit** to continue.

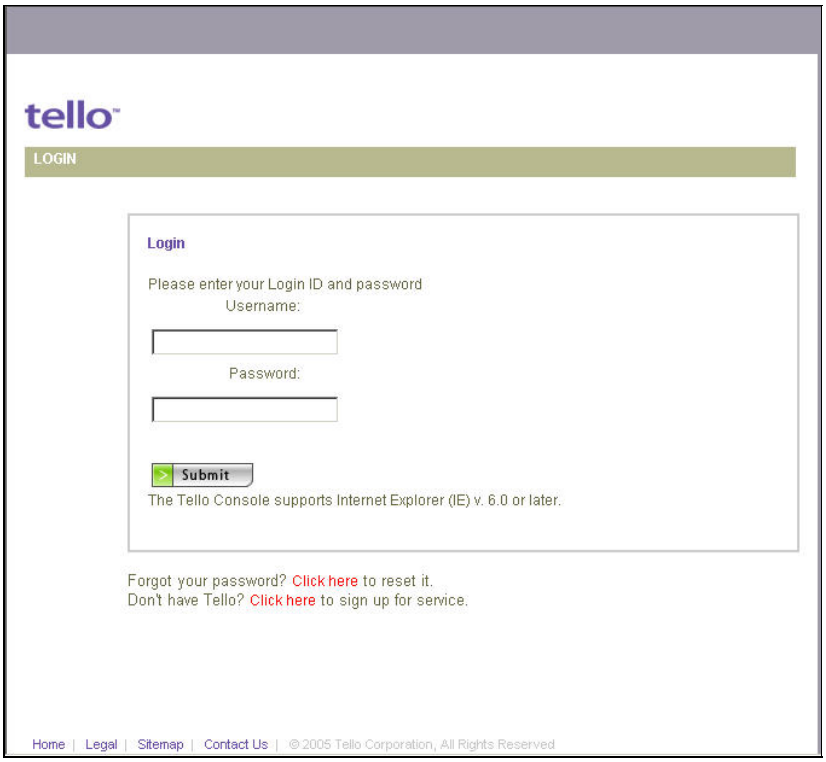

Figure 34: Tello Login

The main page will appear as shown below. After reviewing or making changes to the information provided, logout of the site by selecting the **Logout** link at the top of the page.

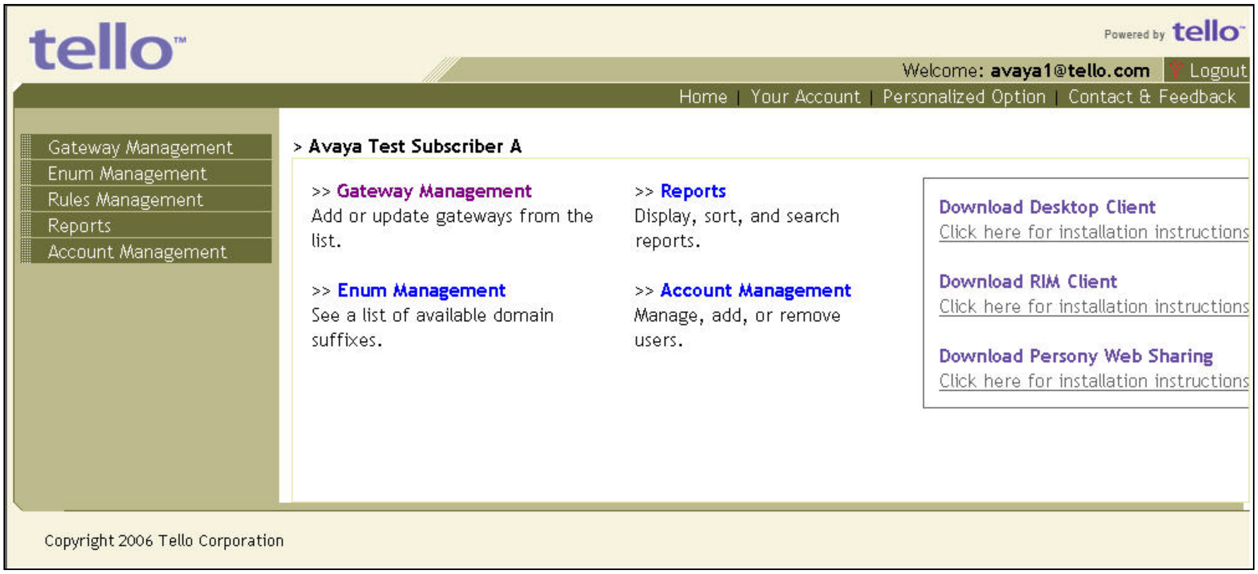

Figure 35: Subscriber Home Page

5.2. Gateway Management

To access information on the SIP proxy (or gateway) associated with this account, select **Gateway Management** from the list in the left-hand pane of the page. The **Gateway Management** page will appear as shown in **Figure 36**.

Figure 36: Gateway Management

To change the gateway information, select the checkbox next to the gateway of interest and select the **Update** button for that entry. This will change the gateway information into a row of editable fields as shown in **Figure 37**. Update any necessary fields and select **Submit**.

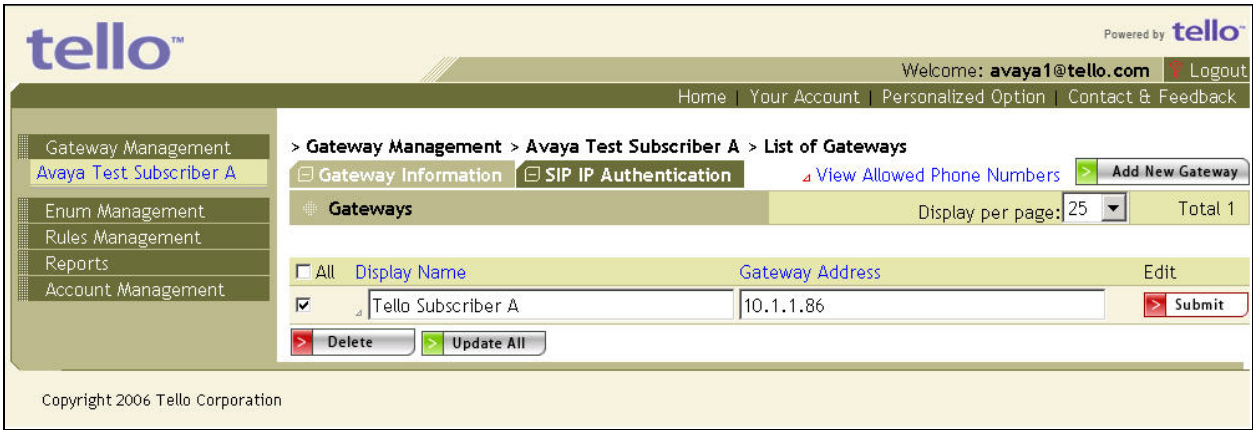

Figure 37: Gateway Management Update

To view the phone numbers associated with this account, select the **View Allowed Phone Numbers** link on the **Gateway Management** page. The list of phone numbers will appear in a separate window as shown in **Figure 38**. The example below shows only the numbers with area code 700 and not the number with area code 732 assigned to this site. Select **close** to close the window.

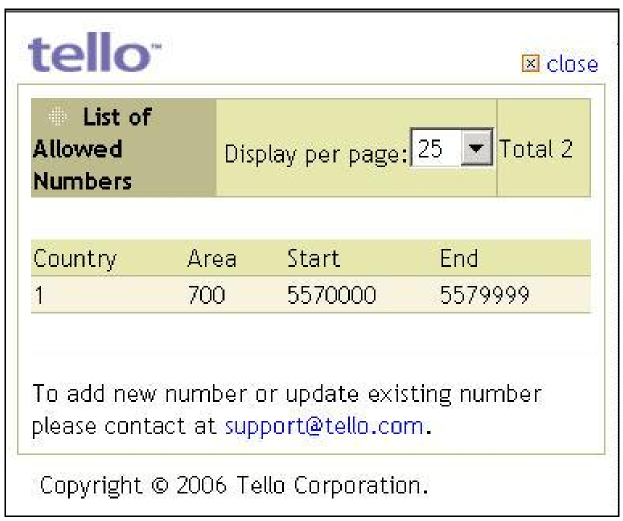

Figure 38: Phone Number List

5.3. Account Management

To access general information on the subscriber account, select **Account Management** from the list in the left-hand pane of the page. The **Account Management** page will appear as shown in **Figure 39**.

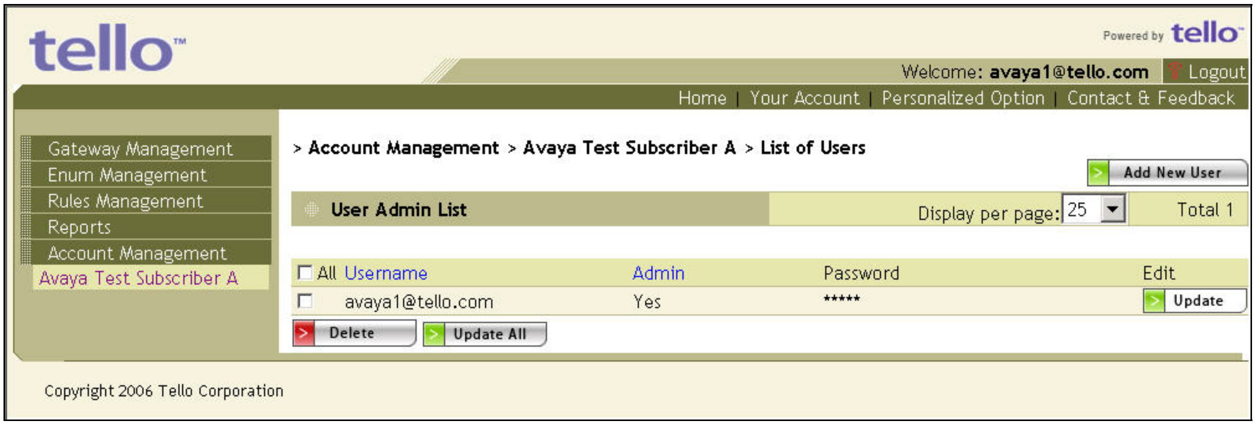

Figure 39: Account Management

To change the general account information, select the checkbox next to the username of interest and select the **Update** button for that entry. This will change the username information into a row of editable fields with a **Submit** button. Update any necessary fields and select **Submit**.

6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the Tello Connect service and an Avaya SIP-enabled IP telephony network. This section covers the general test approach and the test results.

6.1. General Test Approach

Two enterprise sites each containing an Avaya SIP-enabled IP telephony network were connected to the Internet. Each enterprise subscribed to Tello Connect and established a SIP signaling connection between the service and the on-site Avaya SES via SIP trunking. This allowed each enterprise site to place and receive calls to any Tello Connect subscriber via the Internet using SIP. The general test approach was to place and receive calls from each enterprise site to both subscribers and non-subscribers and verify that calls were connected with acceptable voice quality. The parameters of the calls were varied to include different types of endpoints, codecs, dialing strings, etc. Basic serviceability and recovery was tested by placing calls when the Internet connection was down at the near-end or far-end and ensuring the calls were rerouted to the PSTN.

6.2. Test Results

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

- Outgoing calls from the Avaya SIP-enabled IP telephony network to Tello Connect subscribers.
- Outgoing calls from the Avaya SIP-enabled IP telephony network to non-subscribers. These calls were first sent to Tello Connect, could not be completed and then rerouted by Avaya Communication Manager to the PSTN.

- Incoming calls to the Avaya SIP-enabled IP telephony network from Tello Connect subscribers.
- Incoming calls to the Avaya SIP-enabled IP telephony network from non-subscribers and intra-switch calls were made for completeness. However, these calls are not touched by Tello Connect.
- Calls to/from SIP, H.323, digital and analog endpoints in the Avaya enterprise network.
- Various call types including: local, long distance, and toll-free calls.
- Calls using G.711 and G.729B codecs.
- Calls were left up for more than 35 seconds to verify certain SIP protocol timers.
- DTMF transmission using RFC 2833.
- Proper feature operation with hold, transfer and conference.
- Voicemail coverage and retrieval for endpoints at the enterprise site.
- Placing calls with Internet links down at the near-end or far-end. These calls are rerouted by Avaya Communication Manager to the PSTN.

It should be noted that this combined Avaya/Tello solution does not support direct IP-to-IP media (also known as media shuffling) for SIP calls. Direct IP-to-IP media allows IP endpoints to send audio (RTP) packets directly to each other without using media resources on the Avaya Media Gateway. Thus, as described earlier in Section 3.5, it is necessary to disable shuffling on the SIP signaling group.

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 access the Tello Connect service.

1. Verify that endpoints at the Tello subscriber site can place calls to another Tello subscriber and that the call can remain 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 Tello subscriber site can receive calls from another Tello subscriber and that the call can remain active for more than 35 seconds.
3. Verify that the user on the far-end subscriber site can terminate an active call by hanging up.
4. Verify that an endpoint at the near-end subscriber site can terminate an active call by hanging up.
5. Verify that a call to a non-subscriber is successful by routing over the PSTN.
6. Verify the routing of any of the previous calls by using the **status trunk** command on both the SIP and PSTN trunks while the call is active. If the call is routed via Tello Connect, at least one channel will be active on the SIP trunk and none on the PSTN trunk. If the call is routed via the PSTN, at least one channel will be active on the PSTN trunk and none on the SIP trunk.

8. Support

For technical support on Tello Connect, contact Tello via the support link at http://www.tello.com.

9. Conclusion

These Application Notes describe the steps required to configure SIP trunking between the Tello Corporation Tello Connect service and an Avaya SIP-enabled IP telephony network. This solution has passed all necessary compliance testing.

10. References

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

[1] *Administrator Guide for Avaya Communication Manager*, June 2005, Issue 1, Document Number 03-300509.

[2] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, June 2005, Issue 9, Document Number 210-100-500.

[3] *Converged Communications Server R3.0 Installation and Administration Guide (SIP Enablement Services R3.0),* June 2005, Issue 5, Document Number 555-245-705.

[4] *IA 770 INTUITY AUDIX Release 3.0 Installation, Upgrades, and Troubleshooting,* April 2005, Issue 1, Document Number 11-300532.

[5] *SIP Support in Release 3.0 of Avaya Communication Manager Running on the Avaya S8300, S8500, S8500B, S8700, and S8710 Media Server*, June 2005, Issue 5, Document Number 555-245-206.

[6] *4600 Series IP Telephone Release 2.3 LAN Administrator Guide,* November 2005, Issue 2.3, Document Number 555-233-507.

Additional information about Tello Connect is available at http://www.tello.com.

APPENDIX A: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within 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 Avaya SES:

- Normal text characters and numbers match themselves.
- Common metacharacters used are:
  - A period **.** matches any character once (and only once).
  - A asterisk ***** matches zero or more of the preceding characters.
  - Square brackets enclose a list of any character to the 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.
  - Curley brackets containing an integer 'n' indicate that the preceding character must be matched exactly 'n' time. 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
```

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