Avaya Solution & Interoperability Test Lab

Application Notes for the Hitachi Communication Technologies NT-SG with Avaya SIP Enablement Services and Avaya Communication Manager - Issue 1.0

Abstract

These Application Notes describe the configuration of the Hitachi Communication Technologies NT-SG with an Avaya IP Telephony solution consisting of Avaya Communication Manager, Avaya SIP Enablement Services, Avaya 4600 Series H.323 Telephones, Avaya SP-1020A SIP Telephone, and Avaya one-X™ Desktop Edition. The Hitachi Communication Technologies NT-SG is a session border controller that integrates signaling and media control for SIP. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration of the Hitachi Communication Technologies NT-SG in an Avaya IP Telephony environment consisting of an Avaya SIP Enablement Services, Avaya Communication Manager, Avaya 4600 Series H.323 Telephones, Avaya SP-1020A SIP Telephone, and Avaya one-X™ Desktop Edition.

The Hitachi Communication Technologies NT-SG is a session border controller that integrates signaling and media control for SIP. It resides at the enterprise's network border and serves as both the source and destination for all SIP signaling messages and media streams entering or exiting the enterprise's network.

Figure 1 illustrates the configuration that was used to verify the interoperability the Hitachi Telecommunication Technologies NT-SG with an Avaya IP Telephony solution using SIP trunking. The Avaya IP telephony solution located on the customer enterprise site contained:

- Avaya S8400 Media Server installed into an Avaya G600 Media Gateway. The S8400 served as the host processor for Avaya Communication Manager.
- Avaya SIP Enablement Services software operating on an Avaya S8500B server platform.
- Avaya 4600 series IP telephone (configured to use the H.323 protocol).
- Avaya SP-1020A SIP telephone.
- Avaya one-X™ Desktop Edition SIP Softphone.

The SIP Service Provider Network consists of the following SIP components:

- SIP Proxy server
- SIP Registrar server
- SIP redirect server
- Phone Numbers

In the configuration shown in **Figure 1**, one network interface on the Hitachi Communication Technologies NT-SG in the customer enterprise site was connected to the customer enterprise site and the other network interface was connected to the SIP Service Provider network. Two SIP trunks were configured between the Avaya Communication Manager and the Avaya SIP Enablement Services. One SIP trunk was used for the sessions through the Hitachi Communication Technologies NT-SG and the other was used for the sessions from/to the Avaya SP-1020A SIP Telephone and the Avaya one-X™ Desktop Edition. The Avaya SIP Enablement Services recognized the Avaya Communication Manager as a media server, and the Hitachi Communication Technologies NT-SG as a trusted host. The Hitachi Communication Technologies NT-SG terminated the SIP signaling messages and media streams between the customer enterprise site and the SIP Service Provider network.

Telephone numbers are assigned by the SIP telephony service provider. In these Application Notes, two numbers were assigned. One was "050-3380-4036" as a pilot number, which was registered with the Call Agent of the SIP telephony service provider through the SIP REGISTER message. The other was "050-3380-4037" as a branch number, which was not registered with the Call Agent

The administration of the network infrastructure shown in **Figure 1** is not the focus of these Application notes and will not be described. For administration of the network infrastructure shown in **Figure 1** refer to the appropriate documentation listed in **Section 10**.

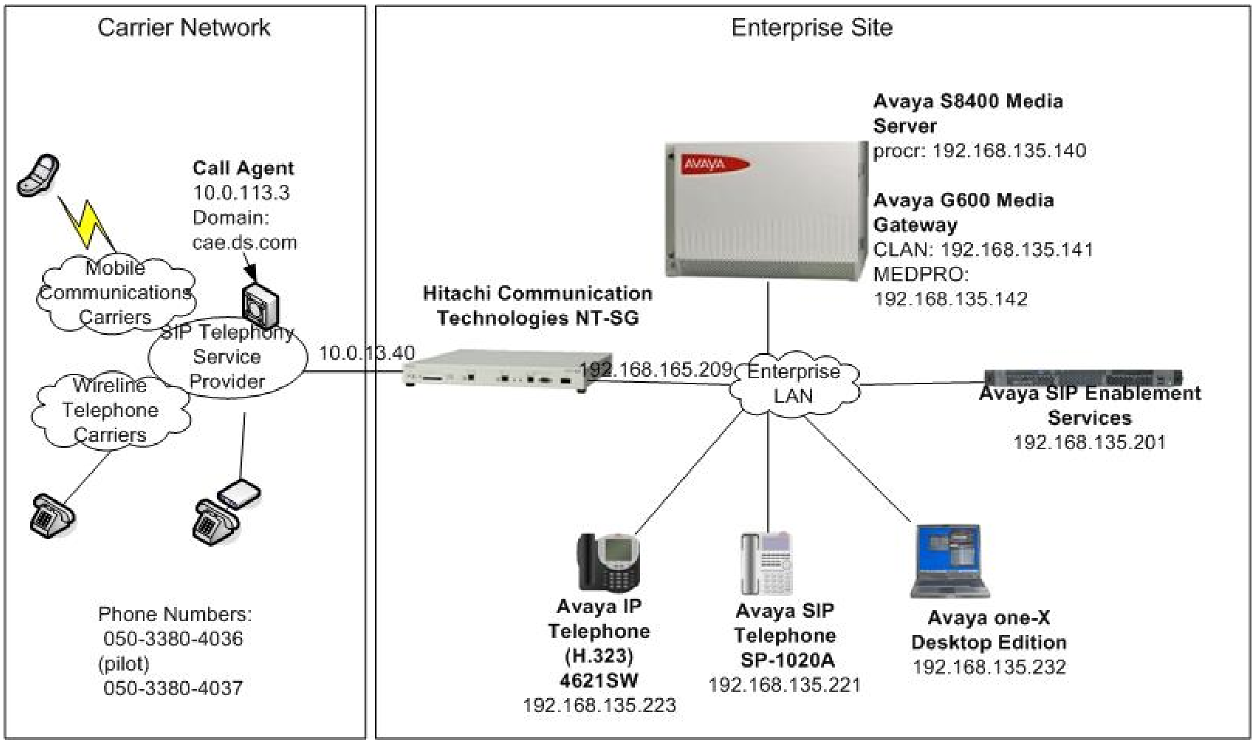

Figure 1: Network Configuration Diagram

2. Equipment and Software Validated

The following equipments and software were used for the configuration in **Figure 1**.

Equipment	Software
Avaya SIP Enablement Services	3.1, load 18.0
Avaya S8400 Media Server	3.1.1, load 628.7
Avaya G600 Media Gateway C-LAN (TN799DP HW0) Media Processor (TN2302AP HW12)	 FW17 FW110
Avaya 4620SW IP Telephone	2.4 (H.323)
Avaya SP-1020A SIP Telephone	2.02.01
Avaya one-X Desktop Edition	R2.1
Hitachi Communication Technologies NT-SG	11-01

Table 1: Equipment and Software

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 and RTP voice packets sent to the Hitachi NT-SG.

The SIP trunk also provides the trunking necessary for SIP endpoint devices such as Avaya SP-1020A SIP telephone and Avaya one-X Desktop Edition to use Avaya Communication Manager in the recommended OPS configuration. It is not necessary to have SIP endpoints in order to use SIP trunking to the Hitachi NT-SG. Thus the steps for SIP endpoints administration are omitted in these Application Notes.

SIP signaling messages for all incoming calls from the SIP Service Provider Network to the enterprise network are terminated at the Hitachi NT-SG, which forwards them to the Avaya SES and are routed to Avaya Communication Manager via the SIP trunk. All outgoing calls to the SIP Service Provider network via the Hitachi NT-SG are routed through Avaya Communication Manager in order to use features such as automatic route selection and class of service restrictions. Avaya Communication Manager creates the outbound SIP signaling that is routed via Avaya SES to the Hitachi NT-SG.

The following configuration was performed via the System Access Terminal (SAT).

The steps shown in this section were configured on the Avaya S8400 media Server at the enterprise site.

Step 1 Confirm Necessary Optional Features

Using the SAT, verify that there exist sufficient SIP Trunks and Off-PBX Telephones capacities by displaying the **System-Parameters Customer-Options** from shown in **Figure 2**. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

On Page 1 of the **System-Parameters Customer-Options** form, verify that the number of OPS stations available is sufficient for the number of SIP telephones to be used.

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V13
       Location: 2                            RFA System ID (SID): 1
       Platform: 20                           RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 1300  170
                                      Maximum Stations: 900   53
                              Maximum XMOBILE Stations: 100   5
                    Maximum Off-PBX Telephones - EC500: 20    1
                    Maximum Off-PBX Telephones -   OPS: 20    6
                    Maximum Off-PBX Telephones - SCCAN: 0     0



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

Figure 2: System-Parameters Customer-Options Form – Page 1

On Page 2, verify that the number of SIP trunks supported by the system is sufficient for the combination of trunks to the Hitachi NT-SG, SIP endpoints and any other SIP trunks used.

```
display system-parameters customer-options                        Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 100   22
           Maximum Concurrently Registered IP Stations: 100   1
             Maximum Administered Remote Office Trunks: 400   0
Maximum Concurrently Registered Remote Office Stations: 900   0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 10    0
                  Maximum Video Capable H.323 Stations: 40    0
                   Maximum Video Capable IP Softphones: 40    6
                       Maximum Administered SIP Trunks: 400   64

   Maximum Number of DS1 Boards with Echo Cancellation: 1     1
                             Maximum TN2501 VAL Boards: 2     0
                    Maximum G250/G350/G700 VAL Sources: 0     0
           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 3: System-Parameters Customer-Options Form – Page 2

Step 2 Assign Node Names

In the **IP Node Names** form, assign the node name and IP address for Avaya SIP Enablement Services at the enterprise site. In this case “SES-01” and “192.168.135.201” are being used, respectively. The SES node name will be used throughout the other configuration screen of Avaya Communication Manager. In this example “procr” and “192.168.135.140” are the name and IP address assigned to the S8400 Media Server, and “shib-cln1” and “192.168.135.141” are the name and IP address assigned to the C-LAN in the G600 Media Gateway.

```
change node-names ip                                              Page   1 of   1
                                  IP NODE NAMES
    Name              IP Address          Name                 IP Address
SES-01              192.168.135.201                              .   .   .
Shib-prw1           192.168.135.142                              .   .   .
default             0  .0  .0  .0                                .   .   .
ia770               192.168.135.147                              .   .   .
nagasaki            192.168.131.170                              .   .   .
procr               192.168.135.140                              .   .   .
shib-cln1           192.168.135.141                              .   .   .
shib-val1           192.168.135.144                              .   .   .
                       .   .   .                                 .   .   .
                       .   .   .                                 .   .   .
                       .   .   .                                 .   .   .
                       .   .   .                                 .   .   .
                       .   .   .                                 .   .   .
                       .   .   .                                 .   .   .
                       .   .   .                                 .   .   .
                       .   .   .                                 .   .   .
( 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 4: IP Nodes Name Form

Step 3 Define IP Network Region

In the **IP Network Region** form, define the parameters associated with the SIP trunk group serving the Avaya SES proxy.

- The **Authoritative Domain** field is configured to match the domain name configured on the Avaya SES. In this configuration, the domain name is *alj.apac.avaya.com*.
- By default, **IP-IP Direct Audio** is enabled to allow audio traffic to be sent directly between SIP endpoints without using media resources in the Avaya G600 Media Gateway.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within IP network region 1. In this configuration, this codec set will apply to calls with the Hitachi NT-SG as well as any IP phone (H.323 or SIP) within the enterprise.

In this case, the SIP trunk is assigned to the same IP network region as the G600 Media Gateway.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: alj.apac.avaya.com
    Name: SIP
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 6999
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        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 5: IP Network Region Form

Step 4 Define IP Codecs

In the **IP Codec Set** form, define the ip-codec value specified in the IP Network Region from **Figure 5**. Although multiple codecs can be listed in priority order in this form, only G.711MU is shown in this case because the Hitachi NT-SG supports only G.711 mu-law.

```
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 6: IP Codec Set Form

Step 5 Configure Signaling Group

Configure the **Signaling Group** form using the "add signaling-group" command to add a new signaling group for SIP trunk between the Avaya Communication Manager and the Avaya SES server.

- 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 SES.
- Specify the node names (i.e. *shib-cln1* and *SES-01*) assigned to the C-LAN board of the Avaya G600 Media Gateway and the Avaya SES in **Figure 4** of **Step 2** in the **Near-end Node Name** field and **Far-end Node Name** field, respectively.
- 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 domain name of Avaya SES in the **Far-end Domain** field. In this configuration, it is *alj.apac.avaya.com*. This domain is specified in the Uniform Resource Identifier (URI) of the "SIP To Address" in the SIP INVITE message.
- **Direct IP-IP Audio Connections** field must be set to *n*. Because the Hitachi NT-SG cannot create any audio connections internally.

```
add signaling-group 50                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 50             Group Type: sip
                        Transport Method: tls








   Near-end Node Name: shib-cln1            Far-end Node Name: SES-01
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region:
      Far-end Domain: alj.apac.avaya.com

                                            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 7: Signaling Group Form

Step 6   Configure Trunk Group

Configure the Trunk Group form using the "add trunk-group" command to add a new trunk group for the SIP trunk between the Avaya G600 Media Gateway and the Avaya SES server. On Page 1 of this form:

- Set the **Group Type** field to *sip*.
- 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 7** of **Step 5**.
- Specify the **Number of Members** supported by this SIP trunk group.

```
add trunk-group 50                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 50                   Group Type: sip          CDR Reports: y
  Group Name: NTT-C test SIP SES-01       COR: 1        TN: 1        TAC: #50
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                          Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 50
                                                     Number of Members: 20
```

Figure 8: Trunk Group Form – Page 1

On Page 3 of this form:

- Set the **Numbering Format** field to *public*.

```
change trunk-group 50                                            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 9: Trunk Group Form – Page 3

Step 7 Configure Calling Party Number Information

Configure the **Numbering Public/Unknown Format** form to send the full calling party number to the far-end.

This case assumes that:

- The SIP telephony service provider assigns two telephone numbers, 050-3380-4036 and 050-3380-4037, to the enterprise site.
- All stations in the enterprise site have a 4-digit extension beginning with 20, 25 or 27.
- All outbound calls use SIP trunk group #50.

```
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

 4 2                                 4
 4 20      50        05033804037     11
 4 25      50        05033804036     11
 4 27      50        05033804037     11
```

Figure 10: Numbering Public/Unknown Format Form

Step 8 Configure Route Pattern

Configure the **Route Pattern** form to route calls to the SIP trunk.

In the examples used in these Application Notes, the enterprise site, which was a subscriber of the SIP telephony service provider was assigned two telephone numbers beginning with 0 as well as other subscribers of the service provider. And all subscribers of the mobile communications SIP Service Providers and wireline telephone SIP Service Providers were assigned a telephone number beginning with 0 as well. In other words, all incoming calls from the SIP Service Provider network via the Hitachi NT-SG to the Avaya SES had a called number beginning with 0, and most of the outgoing calls from the stations to the SIP Service Provider network had a dialed number beginning with 0. Therefore, in order to simplify the address mapping on the Avaya SES, "*9*" is pre-pended to the dialed digits by using the **Inserted Digits** field.

This inserted digit is stripped by the Hitachi NT-SG when it sent the call to the SIP Service Provider network.

```
change route-pattern 50                                         Page   1 of   3
                    Pattern Number: 50  Pattern Name: toSES
                             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: 50   0                        9                                     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
```

Figure 11: Route Pattern Form

Step 9 Configure Incoming Digit Translation

Configure the **Incoming Call Handling Treatment** form to map incoming DID calls to the proper extension.

In the examples used in these Application Notes, the incoming numbers assigned by the SIP telephone service provider do not have a direct correlation to the internal extensions assigned within Avaya Communication Manager. Therefore all incoming called number digits are deleted and replaced by the assigned extension number.

```
change inc-call-handling-trmt trunk-group 50                    Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/       Called    Called        Del  Insert
 Feature         Len      Number
 tie              11  05033804036       all  2501
 tie              11  05033804037       all  2701
```

Figure 12: Incoming Call Handling Treatment Form

Step 10 Save Avaya Communication Manager Changes

Enter “save translations” to make the changes permanent.

4. Configure Avaya SIP Enablement Services

The following steps describe the configuration of the Avaya SIP Enablement Services (SES) to route calls between the SIP Service Provider network and the customer enterprise site through the Hitachi NT-SG. It is assumed that Avaya SES software and the license file have already been installed on Avaya SES. For additional information on these installation tasks, refer to [3].

Step 1 Log into Avaya SIP Enablement Services

Access the Avaya SES Administration web interface, by entering *http://<ip-addr>/admin* as the URL in a web browser, where *<ip-addr>* is the IP address of Avaya SES.

Select the **Launch Administration Web Interface** link from the main screen shown after logged in to the Avaya SES. The Avaya SES administration home screen shown in **Figure 13** should be displayed.

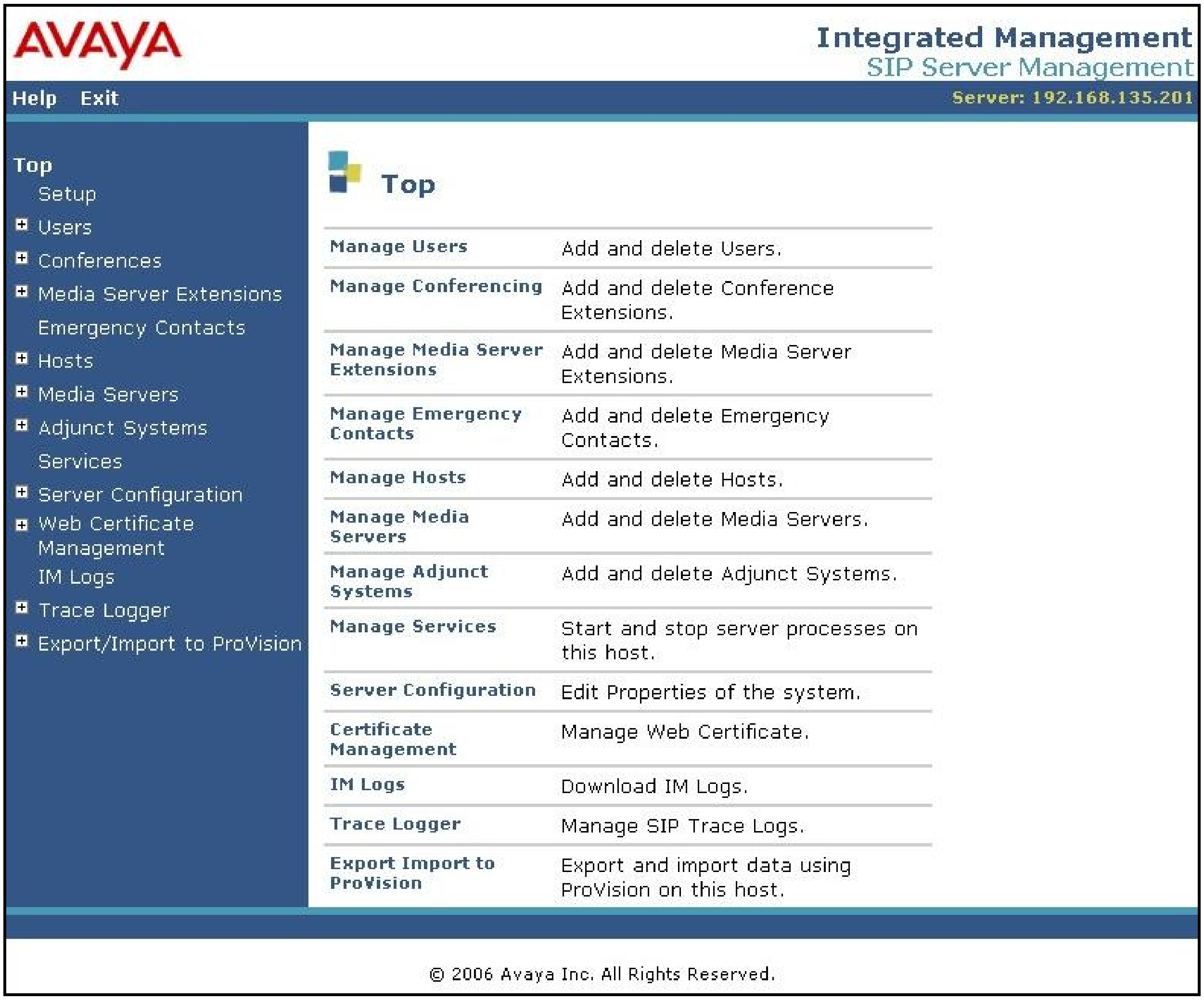

Figure 13: Avaya SES Administration Home Screen

Step 2 Define System Properties

From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**.

- Enter the **SIP Domain** name assigned to Avaya SES. This is the same name that was entered in the **Far-end Domain** field shown in **Figure 7**.
- Enter the IP Address of the Avaya SIP Enablement Services server in **License Host** field as the WebLM application is running on it.
- After configuring the **System Properties** screen, click the **Update** button.

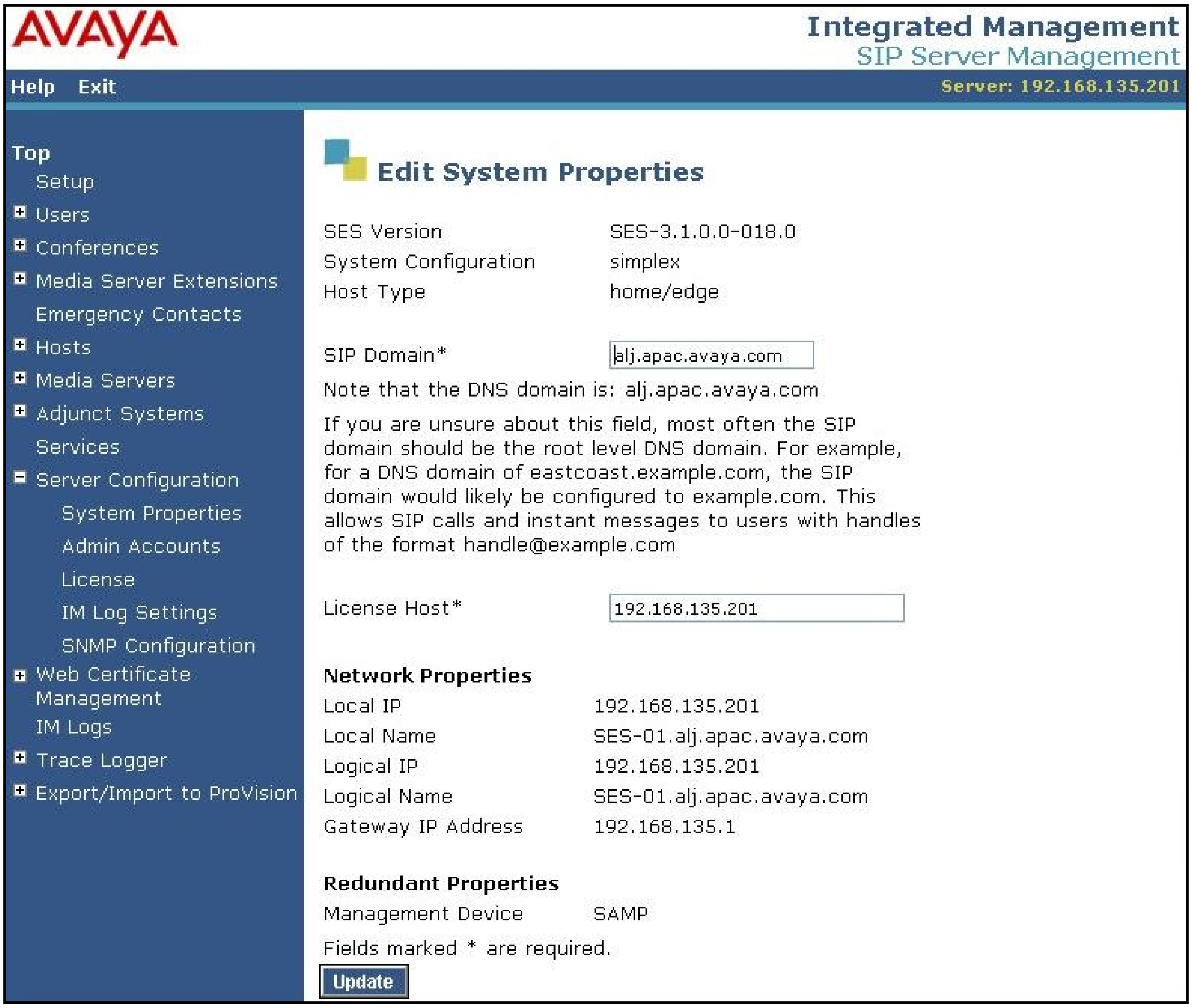

Figure 14: System Properties Screen

Step 3 E

Step 4 nter Avaya SES Host Information

Create a host computer entry for the Avaya SES. The following example shows the **Edit Host** screen since the host had already been added to the system.

From the left pane, click the **Hosts** link and then click **edit** option under the **Commands** section of the subsequent page.

- Enter the **Logical IP** or **Logical Name** (shown in **Figure 14**) of this server in the **Host IP Address** field.
- Enter the **DB Password** that was specified while running the **initial_setup** script during the system installation.
- Configure the **Host Type** field. In this example, the host server was configured as a *home/edge*.
- The default values for the other fields may be used.
- Click the **Update** button.

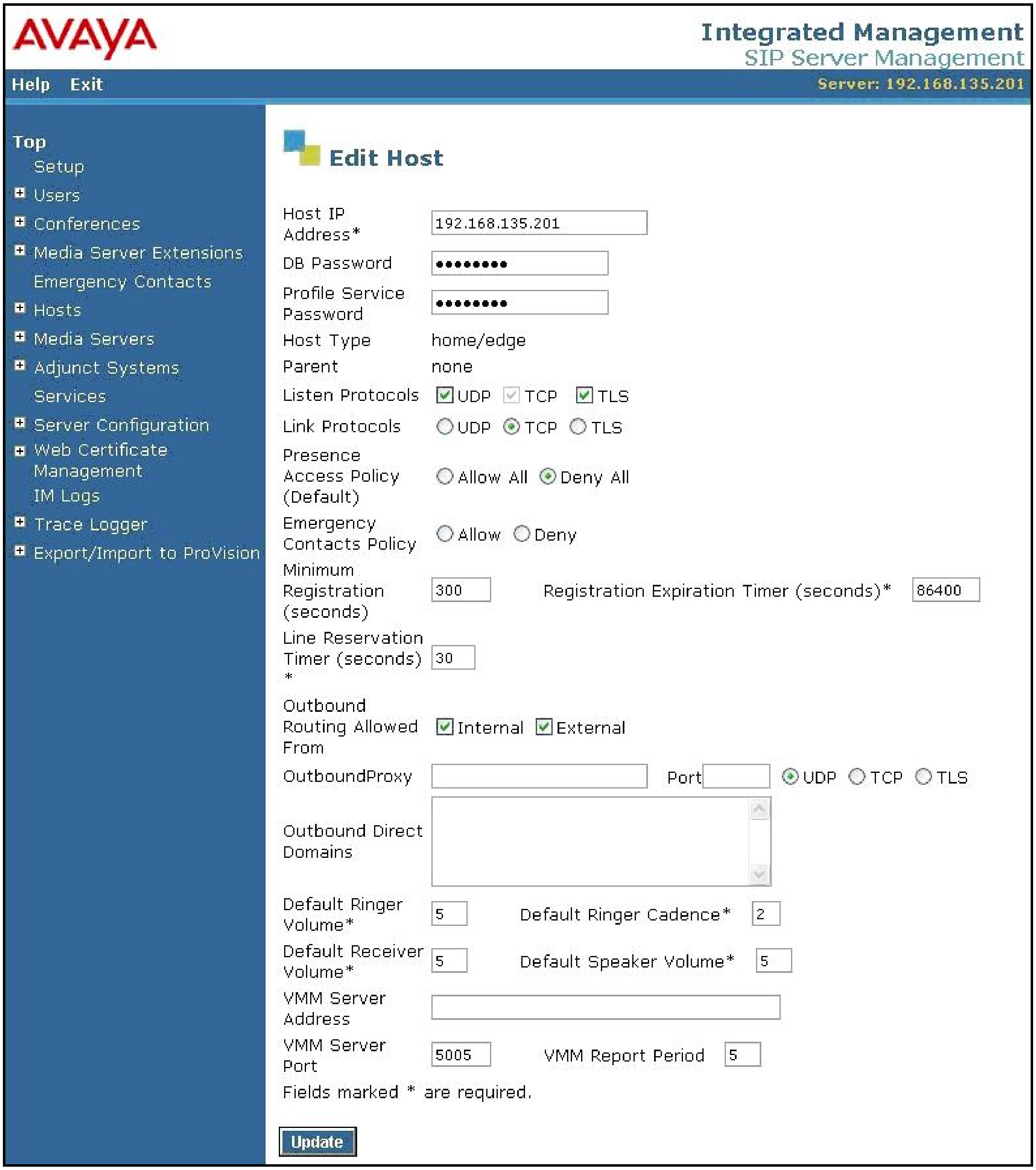

Figure 15: Edit Host Screen

Step 5 Add Avaya Communication Manager as Media Server Interface

Under the **Media Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server.

- Enter a descriptive name in the **Media Server Interface Name** field.
- Select the IP Address of the Avaya SES in the **Host** field. This was configured in the previous step.
- Select *TLS* for the **SIP Trunk Link Type**. TLS is the only link protocol that is supported for SIP trunking with Avaya Communication Manager.
- Enter the IP address of the node specified in the **Near-end Node Name** field of the **Signaling Group** form (**Figure 7**) in the **SIP Trunk IP Address** field. In these Application Notes, the node is the C-LAN board in the Avaya G600 Media Gateway.
- After completing the **Add Media Server Interface** screen, click on the **Add** button.

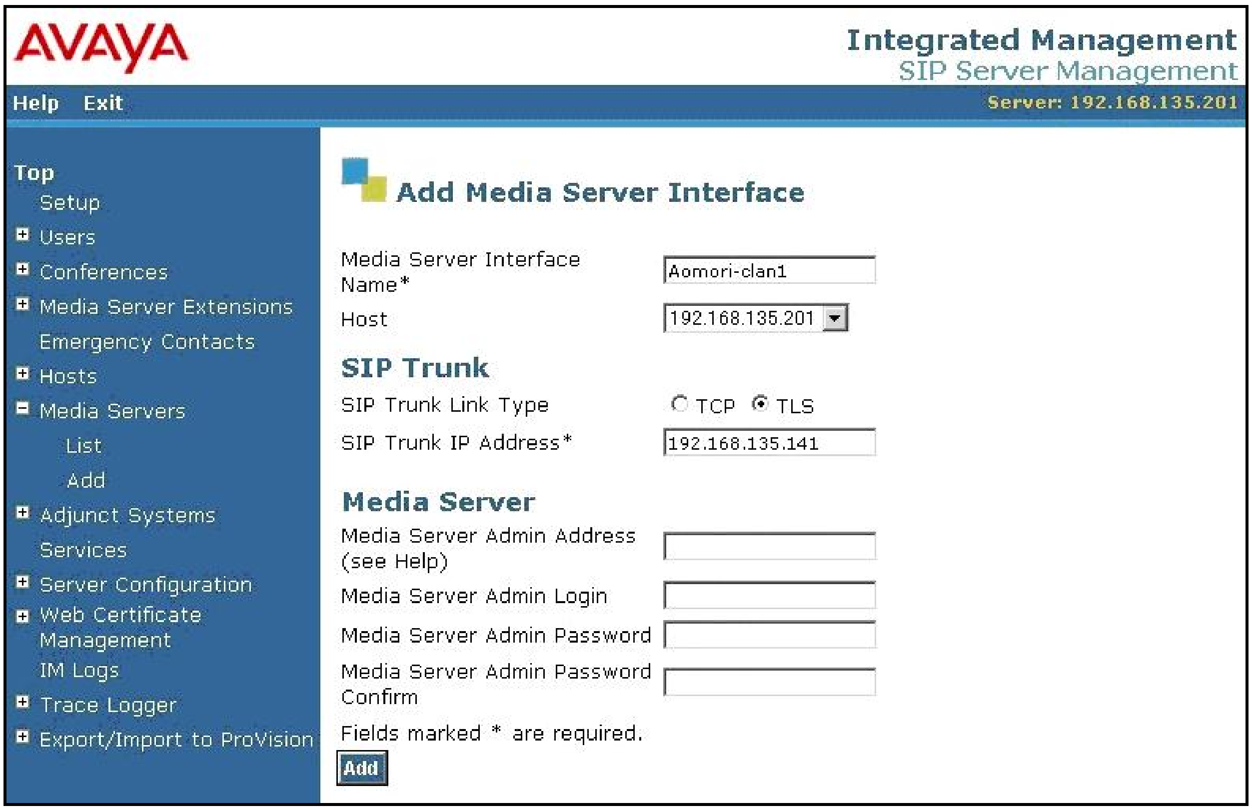

Figure 16: Add Media Server Interface Screen

Step 6 Specify Address Maps to Media Servers

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

This routing compares the Uniform Resource Identifier (URI) of an incoming SIP 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 Managers supported by the Avaya SES server.

In these Application Notes, only incoming calls from the SIP Service Provider network requires 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.

In these Application Notes, all incoming called numbers from the SIP Service Provider network through the Hitachi NT-SG begin with 0.

To minimize the complexity of the address maps, all calls that had the user portion beginning with "0" were routed to the media server.

- 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 17** 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, *^sip:0* means that URIs beginning with "sip:0" will be routed to the C-LAN in the Avaya G600 Media Gateway.
- Leave the **Replace URI** checkbox selected.
- Click the **Add** button once the form is completed.

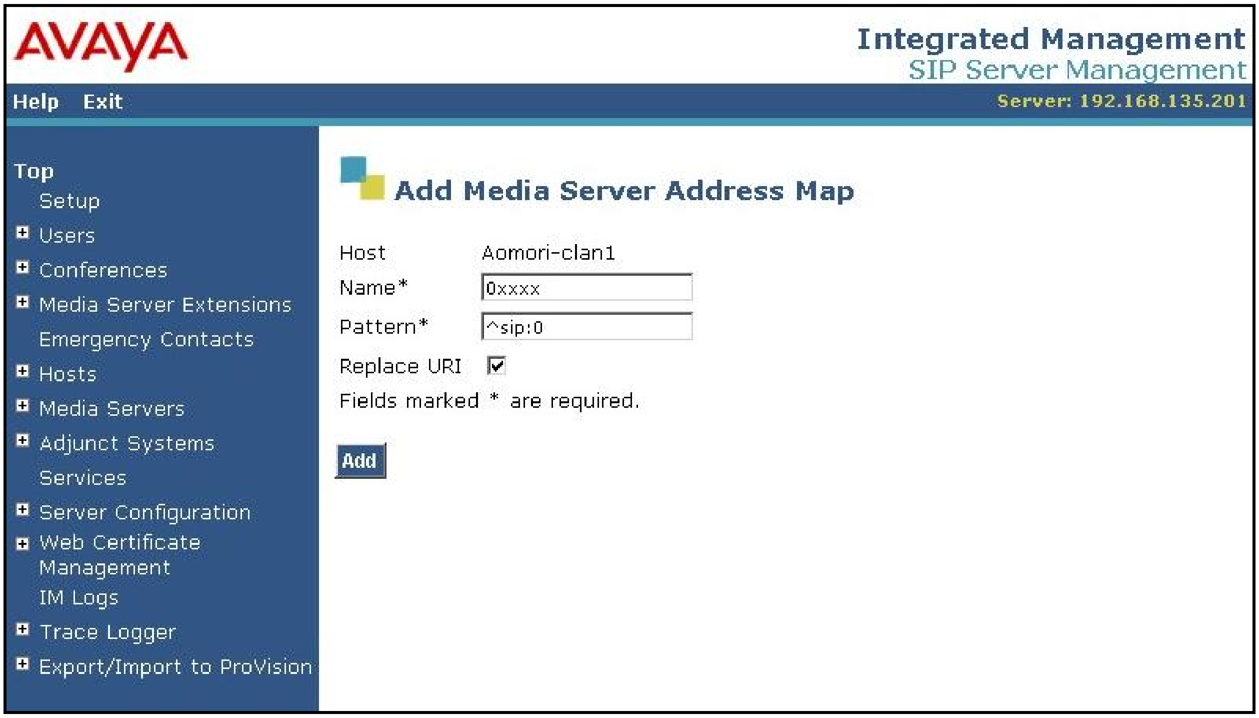

Figure 17: Add Media Server Address Map Screen for Avaya Communication Manager

After configuring the media server address map, the **List Media Server Address Map** screen appears as shown in **Figure 18**.

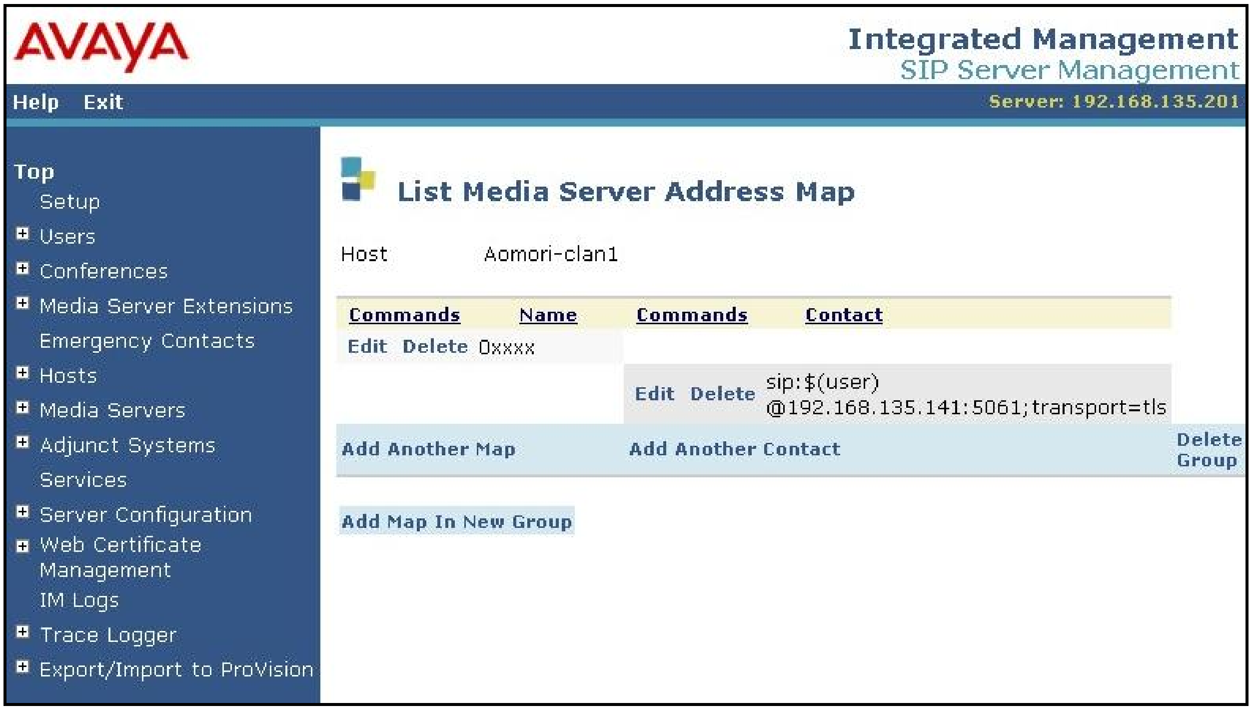

Figure 18: List Media Server Address Map Screen

The **Media Server Contact** is created automatically after the first **Media Server Address Map** is added. The user portion in the original request URI is substituted for `$(user)` of the SIP INVITE message going from Avaya SES to Avaya Communication Manager.

Step 7 Specify Address Maps to the Hitachi NT-SG

As described in **Step 8** of **Section 3**, in this configuration, all outbound calls originated from the extensions had the called number beginning with "9" (regardless of the following digits). Therefore only one dialing pattern of "^sip:9" will be needed.

- Access the **Add Host Address Map** screen by selecting the **Hosts** line in the left pane of the Administration web interface and then clicking on the **Map** link associated with Avaya SES. The **List Host Address Map** screen is displayed.
- Click the **Add Map In New Group** link to display the **Add Host Address Map** screen shown in **Figure 19**.
- 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, *^sip:9* was specified as mentioned above.
- Leave the **Replace URI** checkbox selected.
- Click the **Add** button once the form is completed.

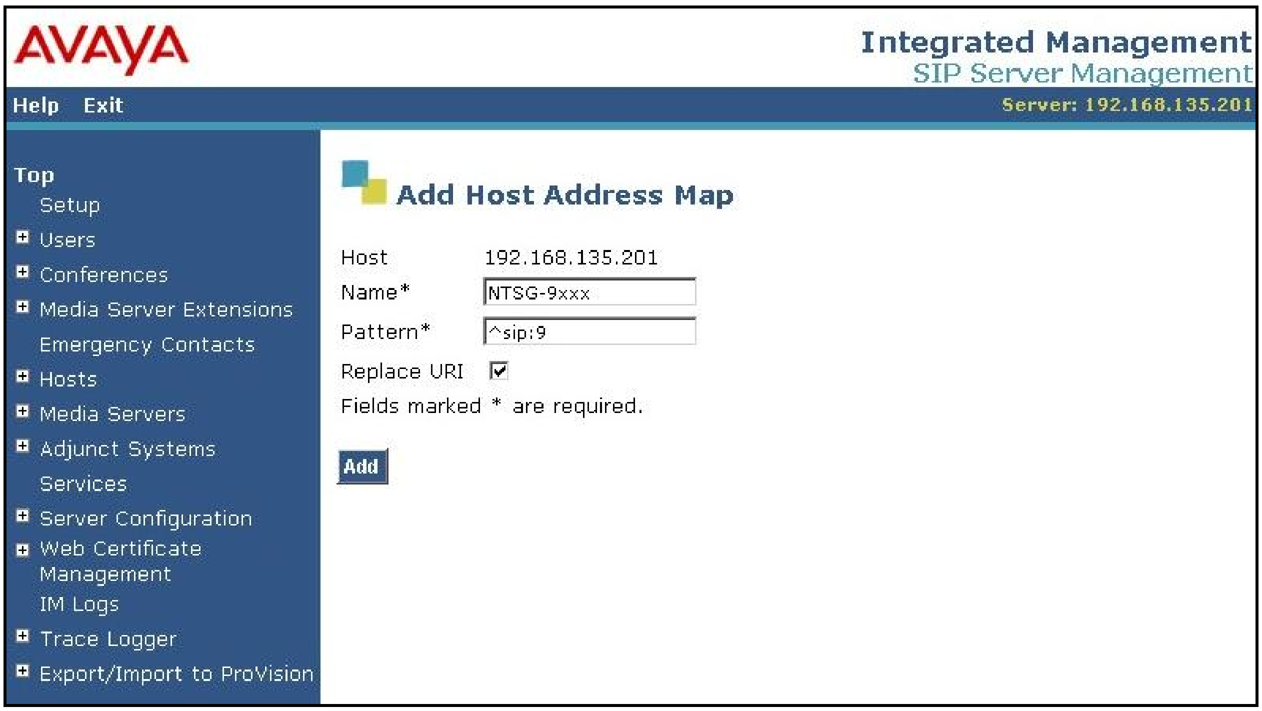

Figure 19: Add Host Address Map Screen

Step 8 Specify the Hitachi NT-SG Host Contact Information

Enter the contact address for the Hitachi NT-SG. In this example, the IP address 192.168.135.209 is used as shown in **Figure 1**.

- As described in **Step 7**, display the List Host Address Map screen.
- Click on the **Add Another Contact** link associated with the address map added in **Figure 19** to open the Add Host Contact screen. In **Figure 20**, the **Contact** field specifies the destination for the call and it is entered as "`sip:$(user)@192.168.135.209;transport=udp`". The user part in the original request URI is substituted for `$(user)`. `transport=udp` is specified since the Hitachi NT-SG supports only UDP as the transport protocol.
- The **Handle** field is the same value as the **Name** field in **Figure 19.**
- Click the **Add** button.

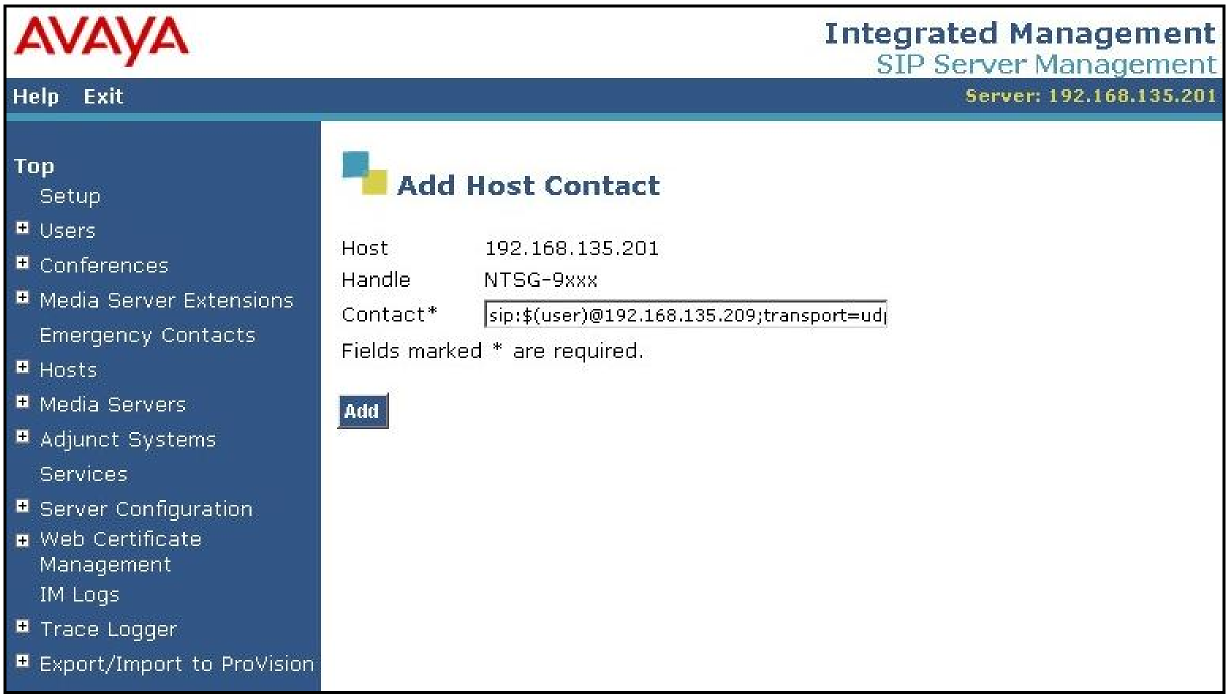

Figure 20: Add Host Contact Screen

After configuring the host address map, the List Host Address Map screen appears as shown in **Figure 21**.

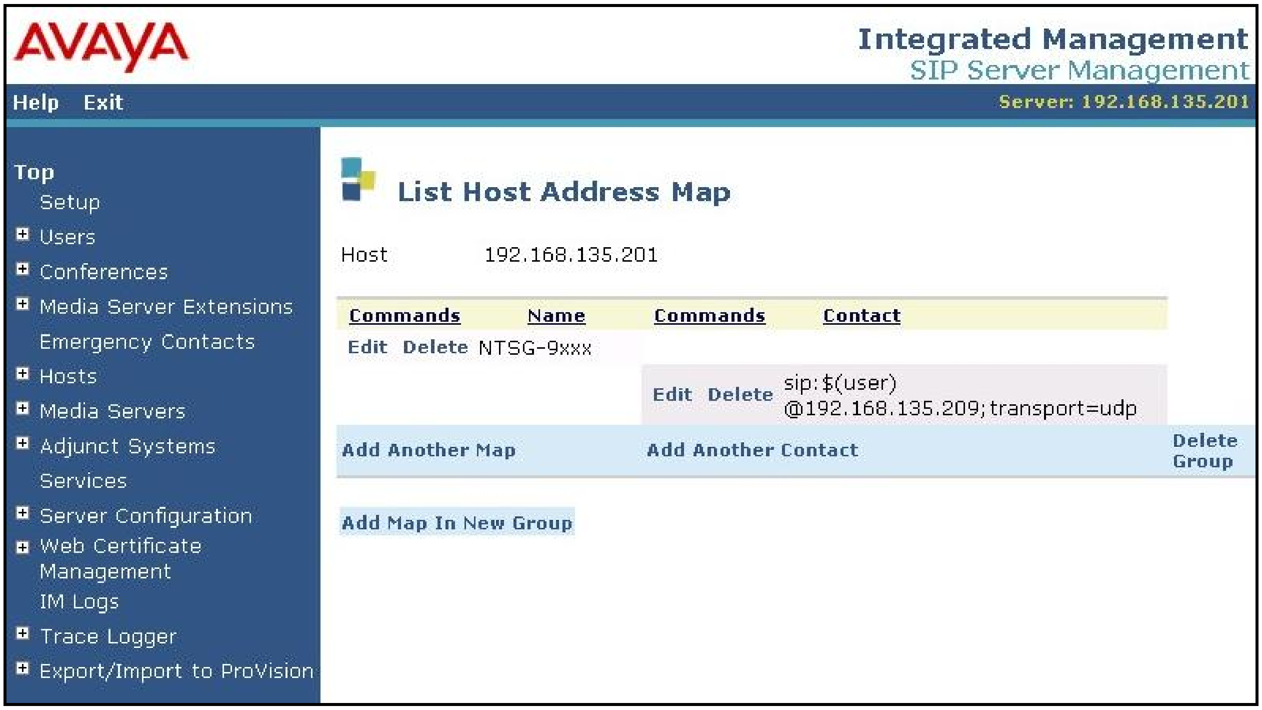

Figure 21: List Host Address Map Screen

Step 9 Save the Changes

Press the **Update** link.

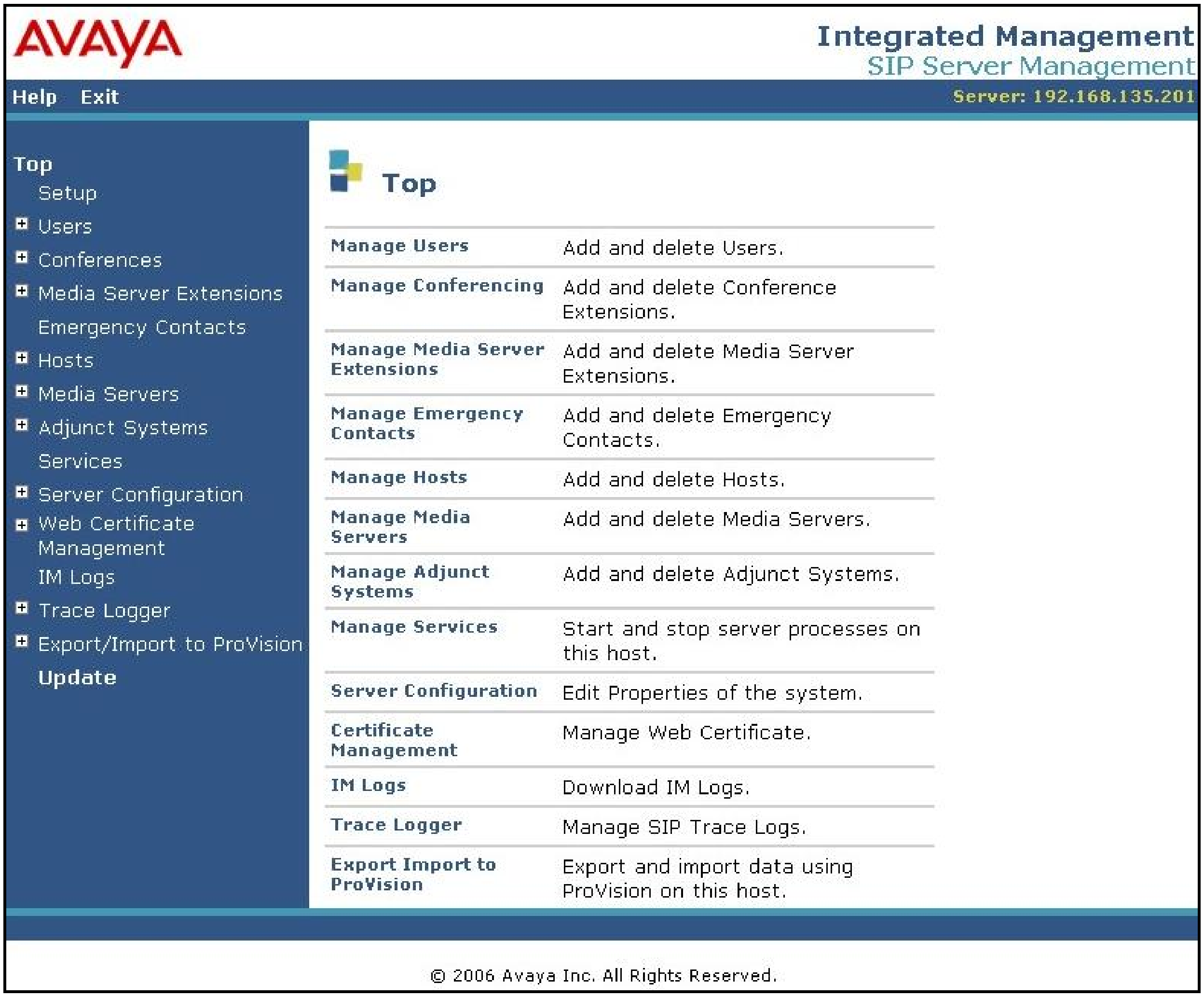

Figure 22: Update Following SES Administrative Changes

Step 10 Specify the Hitachi NT-SG as a Trusted Host

Designate the IP address of the Hitachi NT-SG as a trusted host. As a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the designated IP address.

- Telnet to the Avaya SES IP address and log in using the administrative login and password.
- Enter the following `trustedhost` command at the Linux shell prompt:

  `trustedhost -a 192.168.135.209 -n 192.168.135.201 -c NT-SG`

  The `-a` argument specifies the address to be trusted, `-n` specifies the SES host name, and `-c` adds a comment.
- Verify the entry is correct by the "`trustedhost –L`" command.

- 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 22**.

```
admin@SES-01> trustedhost –a 192.168.135.209 –n 192.168.135.201 –c NT-SG
192.168.135.209 is added to trusted host list.

admin@SES-01> trustedhost -L
Third party trusted hosts.
       Trusted Host       |       CCS Host Name       |         Comment
--------------------------+---------------------------+-------------------------
-
192.168.135.209           | 192.168.135.201           | NT-SG

admin@SES-01>
```

Figure 23: Configuring a Trusted Host

5. Configure Hitachi Communication Technologies NT-SG

This section describes the configuration of the Hitachi NT-SG to function as a session border controller between the SIP Service Provider network and the Avaya SIP Enablement Services (SES) in the enterprise site.

The Hitachi NT-SG is configured using a PC connected to the maintenance port. In these Application Notes, the Hitachi NT-SG is already installed and the maintenance port. For additional information on these installation tasks, refer to [6] and [7].

The SIP implementation of the Hitachi NT-SG is a Back-to-Back User Agent (B2BUA). In addition to the two SIP agents the Hitachi NT-SG also contains an internal control module. The following steps contain the procedure for configuring the SIP agents and control module.

Step 1 Log In

- Telnet to the IP address of the Hitachi NT-SG maintenance port.

The **Main Menu** screen will appear as shown in **Figure 24**.

```
NT-SG ( 32ch)  Module : Main
 Configuration (Main Menu)

                       ---------------------------------
                      | 1 |  Command                     |
                       ---------------------------------
                      | 2 |  Set Configuration           |
                       ---------------------------------
                      | 3 |  Change Configuration        |
                       ---------------------------------
                      | 4 |  Operating Condition         |
                       ---------------------------------
                      | 5 |  Logout                      |
                       ---------------------------------








--------------------------------------------------------------------------------
Selection: Down, Up, Enter
```

Figure 24: Main Menu Screen

Step 2 Configure Network Interface for SIP Modules

This step configures the network interface of two SIP modules in the Hitachi NT-SG.

- Select the **Set Configuration** item by using the down arrow key and press the enter key. The **Set Configuration Module Select** screen will appear as shown in **Figure 25**.

```
NT-SG ( 32ch)  Module : Main
 Configuration (Set Configuration Module Select)

                      ---------------------------------
                     | 1 |  Main Module                 |
                      ---------------------------------
                     | 2 |  SIP Module 1                |
                      ---------------------------------
                     | 3 |  SIP Module 2                |
                      ---------------------------------

--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Enter
```

Figure 25: Set Configuration Module Select Screen

- Select the **Main Module** item and press the enter key. The **Type** form will appear as shown in **Figure 26**.

```
NT-SG ( 32ch)  Module : Main
 Configuration (Type)
                                            Current Configuration :Conf-1
  Configuration Type:[ Conf-1 ]

--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Right, Left, Character, Enter
```

Figure 26: Type Form of Main Module

The main module is capable of storing two configurations. In these Application Notes, **Conf-1** was used for storing the configuration.

- Press the space key until the **Conf-1** type is displayed at the **Configuration Type** field.
- Press the enter key and then the **Conf Menu** screen will appear as shown in **Figure 27**.

```
NT-SG ( 32ch)  Module : Main
 Configuration (Conf Menu)

                       ----------------------------------
                      | 1 |  System                      |
                       ----------------------------------
                      | 2 |  Channel                     |
                       ----------------------------------












--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Enter
```

Figure 27: Conf Menu Screen of Main Module

- Select the **System** item by using the down and up arrow keys, and press the enter key. The **System Menu** screen will appear as shown in **Figure 28**.

```
NT-SG ( 32ch)  Module : Main
 Configuration (System Menu)

                       ---------------------------------
                      | 1 |  Basic                      |
                       ---------------------------------
                      | 2 |  Option                     |
                       ---------------------------------










--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Enter
```

Figure 28: System Menu Screen

- Select the **Basic** item and press the enter key. The **System Basic1** form will appear.
- Press the enter key once to move to the **System Basic2** form for the Port 1 (**Figure 29**).

In these Application Notes, the Port 1 of the Hitachi NT-SG connected to the Avaya SES through the network of the enterprise site, and the Port 2 connected to the SIP Service Provider network. Thus, in the Port 1 screen, the network interface information for the Hitachi NT-SG assigned in the enterprise site was configured.

- Specify the IP address assigned to the Hitachi NT-SG in the enterprise site into the **IP Address** field.
- Specify the subnet mask of the enterprise site network into the **Subnet Mask** field.
- Specify the default gateway address of the enterprise site network into the **Primary** field. In this example, the secondary default gateway was not used. If needed, also the **Secondary** field would be specified.
- Press the enter key once to display the **System Basic3** form for the Port 2 (**Figure 30**).

```
NT-SG ( 32ch)  Module : Main
 Configuration (System Basic2)

  Port1
   IP Address                :[192.168.135.209]
   Subnet Mask               :[255.255.255.0  ]
   Default Gateway
    Primary                  :[192.168.135.1  ]
    Secondary                :[               ]
   SIP IP Header Type Of Service(Hex)      :[00]
   RTP/RTCP IP Header Type Of Service(Hex) :[00]
   VIDEO RTP/RTCP IP Header Type Of Service(Hex) :[00]
   LAN Speed      :[    AUTO    ]









-------------------------------------------------------------------------------
Selection: Tab, Down, Up, Right, Left, Character, Enter
```

Figure 29: System Basic2 Form – for Port 1

In the **System Basic3** form (**Figure 30**), the network interface information for the Hitachi NT-SG assigned from the SIP telephony service provider was configured.

- Specify the IP address assigned to the Hitachi NT-SG in the SIP Service Provider network into the **IP Address** field.
- Specify the subnet mask of the SIP Service Provider network into the **Subnet Mask** field.
- Specify the default gateway address of the SIP Service Provider network into the **Primary** field. In this example, the secondary default gateway was not used. If needed, also the **Secondary** field would be specified.
- Press the enter key three times (the tab key twice) to return to the **System Menu** screen (**Figure 28**). The enter key functions as moving the next screen, and the tab key functions as going back the previous screen.
- Return to the **Type** form (**Figure 26**) using the tab key (press it twice).
- Press the enter key and enter "y" to save the changes to the Hitachi NT-SG.

```
NT-SG ( 32ch)  Module : Main
 Configuration (System Basic3)

  Port2
   IP Address                :[10.0.13.40     ]
   Subnet Mask               :[255.255.255.0  ]
   Default Gateway
    Primary                  :[10.0.13.1      ]
    Secondary                :[               ]
   SIP IP Header Type Of Service(Hex)      :[00]
   RTP/RTCP IP Header Type Of Service(Hex) :[00]
   VIDEO RTP/RTCP IP Header Type Of Service(Hex) :[00]
   LAN Speed      :[    AUTO    ]










--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Right, Left, Character, Enter
```

Figure 30: System Basic3 Form – for Port 2

Step 3   Configure Hitachi NT-SG SIP Module Connected with Avaya SES

This step configures the Hitachi NT-SG SIP module connected with the Avaya SES. This step starts from the **Set Configuration Module Select** screen (**Figure 25**).

- Select the **SIP Module 1** item and press the enter key. The **Type** screen will appear as shown in **Figure 31**.

```
NT-SG ( 32ch)  Module : SIP1
 Configuration (Type)
                                             Current Configuration : Conf-1
  Configuration Type:[ Conf-1 ]              Current User Table    : User-1
                                             Current No.Plan       : No.Plan1
                                             Current Calling Table : Calling1

--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Right, Left, Character, Enter
```

Figure 31: Type Form of SIP Module

Each SIP module is capable of storing two Configurations, two User Tables, two No.Plans and two Calling Tables. In these Application Notes, for the SIP Module 1 the **Conf-1** was configured as the Configuration, and the others were not configured.

- Press the space key until the **Conf-1** type is displayed at the **Configuration Type** field.
- Press the enter key and then the **Conf Menu** screen will appear as shown in **Figure 32**.
- Select the **SIP** item by using the down and up arrow keys, and press the enter key. The **SIP Menu** screen will appear as shown in **Figure 33**.

```
NT-SG ( 32ch)  Module : SIP1
 Configuration (Conf Menu)

                       ---------------------------------
                      | 1 |  SIP                          |
                       ---------------------------------
                      | 2 |  DNS                          |
                       ---------------------------------
                      | 3 |  Numbering                    |
                       ---------------------------------
                      | 4 |  Option                       |
                       ---------------------------------

--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Enter
```

Figure 32: Conf Menu Screen of SIP Module

```
NT-SG ( 32ch)  Module : SIP1
 Configuration (SIP Menu)

                       ---------------------------------
                      | 1 |  SIP Common                   |
                       ---------------------------------
                      | 2 |  SIP User                     |
                       ---------------------------------
                      | 3 |  SIP Filtering                |
                       ---------------------------------

--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Enter
```

Figure 33: SIP Menu Screen

- Select the **SIP Common** Menu option and press the enter key. The **SIP Common1** form will appear as shown in **Figure 34**.
- Specify the IP address of the Avaya SES to the **1st IP Address** field.
- Return to the **Type** form (**Figure 31**) by using the tab key.
- Press the enter key and enter “y” to save the changes to the Hitachi NT-SG.

```
NT-SG ( 32ch)  Module : SIP1
 Configuration (SIP Common1)

 SIP Server
  SIP Server                 :[ Used ]
  1st IP Address             :[192.168.135.201]
  1st Domain Name            :[                                ]
  1st Port                   :[5060 ]
  1st DNS Domain Name        :[                                ]
  2nd IP Address             :[               ]
  2nd Domain Name            :[                                ]
  2nd Port                   :[5060 ]
  2nd DNS Domain Name        :[                                ]
  Switch-Over Execute        :[Off]
  Switch-Back Execute        :[Off]
   Switch-back Interval(min) :[3  ]
  Regist Control             :[Regist]
  Pre Regist Removing        :[Off]
  Regist Contact Check mode  :[ Normal ]
  Server Service Type        :[Off ]
  Regist Stop Control        :[ Off  ]
   Regist Stop Response      :[401] [403] [404] [   ] [   ] [   ] [   ] [   ]
--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Right, Left, Character, Enter
```

Figure 34: SIP Common1 Form

Step 4 Configure Hitachi NT-SG SIP Module Connected with SIP Service Provider Network

This step configures the Hitachi NT-SG SIP module connected with the SIP Service Provider network, which starts with the **Set Configuration Module Select** screen (**Figure 25**).

- Select the **SIP Module 2** item and press the enter key. The **Type** form will appear as shown in **Figure 35**.

In these Application Notes, for the SIP Module 2 the **Conf-1**, **User-1** and **No.Plan1** were configured. The Calling Table was not configured.

For the Configuration of the SIP module 2, specify the SIP entity information in the SIP Service Provider network, the dialed number treatment manner, and the information to register with the SIP Service Provider network.

- Press the space key until the **Conf-1** type is displayed at the **Configuration Type** field.
- Press the enter key and then the **Conf Menu** screen will appear as shown in **Figure 36**.
- Select the **SIP** item by using the down and up arrow keys, and press the enter key. The **SIP Menu** screen will appear as shown in **Figure 37**.

```
NT-SG ( 32ch)  Module : SIP2
 Configuration (Type)
                                             Current Configuration : Conf-1
  Configuration Type:[ Conf-1 ]              Current User Table    : User-1
                                             Current No.Plan       : No.Plan1
                                             Current Calling Table : Calling1

--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Right, Left, Character, Enter
```

Figure 35: Type Form of SIP Module 2

```
NT-SG ( 32ch)  Module : SIP2
 Configuration (Conf Menu)

                       ----------------------------------
                      | 1 |  SIP                          |
                       ----------------------------------
                      | 2 |  DNS                          |
                       ----------------------------------
                      | 3 |  Numbering                    |
                       ----------------------------------
                      | 4 |  Option                       |
                       ----------------------------------

--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Enter
```

Figure 36: Conf Menu Screen of SIP Module

```
NT-SG ( 32ch)  Module : SIP2
 Configuration (SIP Menu)

                       ---------------------------------
                      | 1 |  SIP Common                  |
                       ---------------------------------
                      | 2 |  SIP User                    |
                       ---------------------------------
                      | 3 |  SIP Filtering               |
                       ---------------------------------









--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Enter
```

Figure 37: SIP Menu Screen

- Select the **SIP Common** and press the enter key. The **SIP Common1** form will appear as shown in **Figure 38**. In this form, configure the SIP server (i.e. the Call Agent of the SIP telephony service provider) information.
- Specify the IP address of the Call Agent provided by the SIP telephony service provider to the **1st IP Address** field.
- Specify the domain name provided by the SIP telephony service provider to the **1st Domain Name** field.

```
NT-SG ( 32ch)  Module : SIP2
 Configuration (SIP Common1)

 SIP Server
  SIP Server                :[ Used ]
  1st IP Address            :[10.0.113.3     ]
  1st Domain Name           :[cae.ds.com                      ]
  1st Port                  :[5060 ]
  1st DNS Domain Name       :[                                ]
  2nd IP Address            :[               ]
  2nd Domain Name           :[                                ]
  2nd Port                  :[5060 ]
  2nd DNS Domain Name       :[                                ]
  Switch-Over Execute       :[Off]
  Switch-Back Execute       :[Off]
   Switch-back Interval(min) :[3  ]
  Regist Control            :[Regist]
  Pre Regist Removing       :[Off]
  Regist Contact Check mode  :[ Normal ]
  Server Service Type       :[Off ]
  Regist Stop Control       :[ Off  ]
   Regist Stop Response      :[401] [403] [404] [   ] [   ] [   ] [   ] [   ]
--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Right, Left, Character, Enter
```

Figure 38: SIP Common1 Form

- Press the enter key to move the next screen (**Figure 39**).
- Select the **System** in the **Invite Calling Number Set** field by using the space key. This means that the number registered with the Call Agent by the SIP REGISTER message would be used as the calling number in the SIP INVITE message if the Hitachi NT-SG didn't receive a calling number or received an unknown number from the Avaya SES. The number that was registered with the Call Agent will be specified in the **User-1** (**Figure 43**).
- Select the **After** in the **Called Number Check** field by using the space key. This means that the Hitachi NT-SG treats the called numbers ***after*** the No.Plan is applied to them. The No.Plan will be configured in the screen shown in **Figure 46**.

```
NT-SG ( 32ch)  Module : SIP2
 Configuration (SIP Common2)

 SIP System
  Invite Calling Number Set  :[ System ]
   Privacy Header Scheme Set :[sip]
  Calling Number Notice      :[   Notice   ]
  Invite Arrival Mode        :[    Normal    ]
  Invite Expires Timer(s)    :[300]
  Invite Redirect            :[Off]
  Calling Number Get         :[ Username  ]
  Provisional Path Control   :[On ]
  Invite Arrival Filtering   :[Off]
  Slide Stop Control         :[Off]
  Slide Stop Cause           :[   ] [   ] [   ]
  Called Number Check        :[After]
  Calling Number Control     :[PRE]
  Called Number Mode         :[Request]
  Dial 184/186 Del           :[On ]
  Receive INFO Mode          :[DTMF]
  Authorization Set          :[On ]
  Contact Header Set         :[Original]
--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Right, Left, Character, Enter
```

Figure 39: SIP Common2 Form

- Return to the **SIP Menu** screen (**Figure 37**) by using the tab key.
- Select the **SIP User** and press the enter key. The **SIP User** screen will appear as shown in **Figure 40**.

```
NT-SG ( 32ch)  Module : SIP2
 Configuration (SIP User)

                       ----------------------------------
                      | 1 |  Common                      |
                       ----------------------------------















--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Enter
```

Figure 40: SIP User Screen

- Press the enter key. The **SIP User Common** form will appear as shown in **Figure 41**.
- Specify the user ID and password, which are assigned by the SIP telephony service provider in advance, to register with the Call Agent of the SIP telephony service provider in the **1st User ID** and **1st Password** fields. In this example, the SIP telephony service provider assigned "user" and "passwd" as the user ID and password to the enterprise site.

```
NT-SG ( 32ch)  Module : SIP2
 Configuration (SIP User Common)

 SIP User Common
  System Domain Name          :[                                 ]
  System User Name            :[Gateway                          ]
  System Display Name         :[                                 ]
  System Contact User Name    :[                                 ]
  Non-Notice Domain Name      :[anonymous.invalid                ]
  Non-Notice User Name        :[anonymous                        ]
  Non-Notice Display Name     :[Anonymous                        ]
  Authorization Challenge Data
   1st User ID                :[user                             ]
   1st Password               :[passwd                           ]
   2nd User ID                :[                                 ]
   2nd Password               :[                                 ]
  SDP User Name               :[hitachi                          ]
  SDP Session Name            :[Session SDP                      ]




--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Right, Left, Character, Enter
```

Figure 41: SIP User Common Form

- Return to the **Type** form (**Figure 35**) by using the tab key.
- Press the enter key and enter "y" to save the changes to the Hitachi NT-SG. After that, the **Set Configuration Module Select** screen (**Figure 25**) will appear.

For the User Table of the SIP module 2, specify the telephone numbers assigned by the SIP telephony service provider. In these Application Notes, two numbers were assigned. One was "050-3380-4036" as a pilot number, which was registered with the Call Agent of the SIP telephony service provider through the SIP REGISTER message. The other was "050-3380-4037" as a branch number, which was not registered with the Call Agent.

- Select the **SIP Module 2** item on the **Set Configuration Module Select** screen (**Figure 25**) and press the enter key. The configuration type screen will appear as shown in **Figure 35**.
- Press the space key until the **User-1** type is displayed at the **Configuration Type** field.
- Press the enter key and then the **Menu** screen will appear as shown in **Figure 42**.

```
NT-SG ( 32ch)  Module : SIP2
 User Configuration Info(Menu)

                       ---------------------------------
                      | 1 |  Table                      |
                       ---------------------------------

--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Enter
```

Figure 42: Menu Screen for User Configuration Info of SIP Module

- Press the enter key. The **User Info – Table No. 0001** form will appear as shown in **Figure 43**.
- Press the tab key to move the cursor to the parameter setting area, which is the area between the **Info field** and the **Contact Parameter q** field. The cursor moves between the **Display Screen** line at the bottom of the screen and the parameter setting area whenever the tab key is pressed.
- Select **Used** in the **Info** filed by using the space key to activate this table.
- Select **M** in the **Main** field by using the space key to specify the pilot number in this table.
- Select **On** in the **Regist** field by using the space key. This means that the User Name specified in this table will be registered with the Call Agent in the SIP telephony service provider. Note that only one table must have both the **Used** (in the **Info** field) and the **M** (in the **Main** field).
- Enter the telephone number assigned by the SIP telephony service provider in the **User Name** field.   One more assigned number, "050-3380-4037" will be entered in other table (**Figure 44**).
- Press the tab key to move the cursor to the **Display Screen** field.
- Press the space key until the **Next** will be displayed in the field, and press the enter key. Then the next table screen will appear as shown in **Figure 44**.

```
NT-SG ( 32ch)  Module : SIP2
 User Info (User Info - Table No. 0001)

 Info                       :[Used]
 Main                       :[M]
 Regist                     :[On]
 User Name                  :[05033804036                         ]
 Display Name               :[                                    ]
 Contact User Name          :[                                    ]
 Del Column                 :[0 ]
 Add Number                 :[                                    ]
 Authorization Challenge Data
  1st User ID               :[                                    ]
  1st Password              :[                                    ]
  2nd User ID               :[                                    ]
  2nd Password              :[                                    ]
 Contact Parameter q        :[     ]



  Display Screen :[Next] Table No. :[    ]
--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Right, Left, Character, Enter
```

Figure 43: User Info Form – Table 1

In this table, one more number assigned by the SIP telephony service provider is configured. Note that this number was not the pilot number for the enterprise site.

- Press the tab key to move the cursor to the parameter setting area.
- Select the **Used** in the **Info** filed by using the space key to activate this table.
- Leave the **Main** field blank.
- Select the **Off** in the **Regist** field by using the space key since this number was not registered with the Call Agent of the SIP telephony service provider with the SIP REGISTER message.
- Enter the telephone number assigned by the SIP telephony service provider in the **User Name** field.
- Press the tab key to move the cursor to the **Display Screen** field.
- Press the space key until the **Menu** will be displayed in the field, and press the enter key. Then the **Menu** screen will appear as shown in **Figure 42**.
- To save the changes to the Hitachi NT-SG, press the tab key until the **Type** form will appear. Then press the enter key and enter “y”. After that, the **Set Configuration Module Select** screen (**Figure 25**) will appear.

```
NT-SG ( 32ch)  Module : SIP2
 User Info (User Info - Table No. 0002)

 Info                        :[Used]
 Main                        :[ ]
 Regist                      :[Off]
 User Name                   :[05033804037                     ]
 Display Name                :[                                ]
 Contact User Name           :[                                ]
 Del Column                  :[0 ]
 Add Number                  :[                                ]
 Authorization Challenge Data
  1st User ID                :[                                ]
  1st Password               :[                                ]
  2nd User ID                :[                                ]
  2nd Password               :[                                ]
 Contact Parameter q         :[     ]




  Display Screen :[Next] Table No. :[    ]
--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Right, Left, Character, Enter
```

Figure 44: User Info Form – Table 2

For the No.Plan of the SIP module 2, specify the numbering plan. All dialed numbers conveyed from the Avaya SES to the Hitachi NT-SG begin with "9" as described in **Step 8** of **Section 3**. This configuration was needed to delete the "9" from the dialed numbers before the Hitachi NT-SG treated the numbers.

- Select the **SIP Module 2** item on the **Set Configuration Module Select** screen (**Figure 25**) and press the enter key. The **Type** form will appear as shown in **Figure 35**.
- Press the space key until the **No.Plan1** type is displayed at the **Configuration Type** field.
- Press the enter key and then the **Numbering Plan Menu** screen will appear as shown in **Figure 45**.
- Select the **Table** item by using the down and up arrow keys, and press the enter key. The **Numbering Plan-Table No.001** form will appear as shown in **Figure 46**.

```
NT-SG ( 32ch)  Module : SIP2
 Configuration (Numbering Plan Menu)

                       ----------------------------------
                      | 1 |  Table                        |
                       ----------------------------------
                      | 2 |  Dest Information             |
                       ----------------------------------















--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Enter
```

Figure 45: Numbering Plan Menu Screen

- Press the tab key to move the cursor to the parameter setting area.
- Move to the **Del Column** field of the **Dial** “9” using the arrow keys.
- Enter “1” in the **Del Column** to delete one digit.
- Press the tab key to move the cursor to the **Display Screen** field.
- Press the space key until the **Menu** will be displayed in the field, and press the enter key. Then the **Numbering Plan Menu** screen will appear as shown in **Figure 45**.
- To save the changes to the Hitachi NT-SG, press the tab key until the **Type** form will appear. Then press the enter key and enter “y”. After that, the **Set Configuration Module Select** screen (**Figure 25**) will appear.

```
NT-SG ( 32ch)  Module : SIP2
 Configuration (Numbering Plan-Table No.001-)

  Numbering Plan Information
   Dial   Mode  PRMTR1 PRMTR2  Name    Del Column   Add Number     Route
    1  =  [2],  [1  ], [32 ]  [     ]  [0 ]     [                ]  [LAN ]
    2  =  [2],  [1  ], [32 ]  [     ]  [0 ]     [                ]  [LAN ]
    3  =  [2],  [1  ], [32 ]  [     ]  [0 ]     [                ]  [LAN ]
    4  =  [2],  [1  ], [32 ]  [     ]  [0 ]     [                ]  [LAN ]
    5  =  [2],  [1  ], [32 ]  [     ]  [0 ]     [                ]  [LAN ]
    6  =  [2],  [1  ], [32 ]  [     ]  [0 ]     [                ]  [LAN ]
    7  =  [2],  [1  ], [32 ]  [     ]  [0 ]     [                ]  [LAN ]
    8  =  [2],  [1  ], [32 ]  [     ]  [0 ]     [                ]  [LAN ]
    9  =  [2],  [1  ], [32 ]  [     ]  [1 ]     [                ]  [LAN ]
    0  =  [2],  [1  ], [32 ]  [     ]  [0 ]     [                ]  [LAN ]
    *  =  [2],  [1  ], [32 ]  [     ]  [0 ]     [                ]  [LAN ]
    #  =  [2],  [1  ], [32 ]  [     ]  [0 ]     [                ]  [LAN ]




  Display Screen :[Next] Table No. :[   ]
--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Right, Left, Character, Enter
```

Figure 46: Numbering Plan-Table No.001- Form

Step 5 Activate Changes

This step activates the above changes on the Hitachi NT-SG.

- Press the tab key until the **Main Menu** screen (**Figure 24**) will appear.
- Select the **Change Configuration** item by using the down and up arrow keys, and press the enter key. The **Change** form will appear as shown in **Figure 47**. Information displayed between “from” and “to” means the current configuration.
- Specify all changes, which were done in **Step 2** through **Step 4**. In these Application Notes, using the space key, specify “**Conf-1**” for the **Configuration change from Conf-1 to** field in all sections, “**User-1**” for the **User Table change from User-1 to** field in the **SIP Module 2** section, and “**No.Plan1**” for the **Table Pattern change from No.Plan1 to** field in the **SIP Module 2** section.
- Press the enter key to activate the changes on the Hitachi NT-SG.
- Enter “y” for the confirmation “Change Configuration (y/n) ?”.
- Enter “y” for the confirmation “Change Configuration & System Reboot(y/n) ?”. Then the Hitachi NT-SG starts the system reboot and the telnet connection with the Hitachi NT-SG is disconnected. In a few minutes, the status indicator on the front panel of the Hitachi NT-SG will show “0”.

```
NT-SG ( 32ch)  Module : Main
 Configuration (Change)

  Main Module
    Configuration change from Conf-1   to :[ Conf-1 ]

  SIP Module 1
    Configuration change from Conf-1   to :[ Conf-1 ]
    User Table change from    User-1   to :[  None  ]
    Table Pattern change from No.Plan1 to :[  None  ]
    Calling Table change from Calling1 to :[  None  ]

  SIP Module 2
    Configuration change from Conf-1   to :[ Conf-1 ]
    User Table change from    User-1   to :[ User-1 ]
    Table Pattern change from No.Plan1 to :[No.Plan1]
    Calling Table change from Calling1 to :[  None  ]





--------------------------------------------------------------------------------
Selection: Tab, Down, Up, Right, Left, Character, Enter
```

Figure 47: Change Form

6. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the Hitachi NT-SG and an Avaya SIP-based configuration. This section covers the general test approach ant the test results.

6.1. General Test Approach

The following scenarios were tested using the network configuration diagram shown in **Figure 1**. The VoIP network of NTT Communications was used as the SIP Service Provider network. This allows the enterprise site to make calls to the subscribers of the telephone SIP Service Providers and mobile communications SIP Service Providers, not only the subscribers of the SIP telephony service provider.

- Outgoing calls t were routed properly, and the SIP messages were properly formatted on the both side of the Hitachi NT-SG.
- Incoming calls f were routed to the DID numbers, and the SIP messages were properly formatted on the both side of the Hitachi NT-SG.
- Calls using SIP and H.323 endpoints supported by the Avaya IP telephony solution.
- Various call types including: local, long distance, international, fax and IVR calls.
- DTMF transmission using G.711.
- Voicemail coverage and retrieval for endpoints.
- Call forwarding incoming calls from the SIP Service Provider network to other participants through the SIP Service Provider network, incoming calls from the SIP

Service Provider network to the extensions in the enterprise site, and calls originated in the enterprise site to other participants through the SIP Service Provider network.

- Transferring incoming calls from the SIP Service Provider network to other participants through the SIP Service Provider network, incoming calls from the SIP Service Provider network to the extensions in the enterprise site, and calls originated in the enterprise site to other participants through the SIP Service Provider network.
- Conferencing with the extensions and other participants through the SIP Service Provider network.
- CLIP and CLIR.

6.2. Test Results

All test cases passed. The following issues were noted.

1. The Hitachi NT-SG only supports the G.711 mu-law codec.
2. The Hitachi NT-SG only supports the udp protocol.

7. Verification Steps

This section provides verification steps that may be performed in the field to verify and that the endpoints can place outbound and receive inbound the SIP Service Provider network through the Hitachi NT-SG.

1. Verify that endpoints at the enterprise site can place calls to the SIP Service Provider network and that the call remains active for more than the session refresh timer value of the SIP Service Provider network. 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 SIP Service Provider network and that the call can remain active for more than the session refresh timer value of the SIP Service Provider network.
3. Verify that the user behind the SIP Service Provider network (e.g. the user of a wireline telephone SIP Service Provider, mobile communications SIP Service Provider, or SIP telephony service provider) 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 the Hitachi NT-SG, visit http://www.hitachi-com.com/.

9. Conclusion

These Application Notes describe the configuration steps required to connect an enterprise site consisting of an Avaya SIP-based telephony solution to a SIP Service Provider network via the Hitachi Communication Technologies NT-SG. The Avaya SIP-based telephony solution with the

NT-SG provides enterprise customers with the cost effective converged network by integrating their telecommunication network with their broadband Internet access network.

10. Additional References

10.1. Documentation

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*, May 2006, Issue 2.1, 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] *Installing and Administering SIP Enablement Services Release 3.1*, February 2006, Issue 1.5, Document Number 03-600768.

[4] *Avaya IA 770 INTUITY AUDIX Messaging Application Release 3.1 Administering the S8300 and S8400 Media Servers to work with IA 770*, February 2006, Issue N/A, Document Number 07-600788.

[5] *SIP Support in Release 3.1 of Avaya Communication Manager Running on the Avaya S8300, S8400, S8500 series, and S8700 series Media Server*, February 2006, Issue 6, Document Number 555-245-206.

The following documentation was included with the Hitachi Communication Technologies NT-SG.

[6] *SIP Protocol Converter NT-SG Setup Terminal Operating Manual*, May 2006, Issue 11.0a.

[7] *SIP Protocol Converter NT-SG Installation and Maintenance Manual*, March 2006, Issue 10.0.

Additional information about the Hitachi Communication Technologies NT-SG is available at http://www.hitachi-com.com/.

10.2. Glossary

CLIP - Calling Line Identification Presentation. The feature allows the phone set to display the caller's phone number before you answer it.

CLIR - Calling Line Identification Restriction. The feature allows you to conceal your identity / phone number when you're making a call to other phone.

TLS - Transport Layer Security. TLS is cryptographic protocol which provides secure communication on the Internet, and is the successor to SSL (Secure Socket Layer).

APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by the NT-SG and the Avaya SIP Enablement Services at the enterprise site. Customers may use these INVITE message for comparison and troubleshooting purposes. Differences in these messages may indicate different configuration options selected.

Sample SIP INVITE Message from NT-SG to Avaya SIP Enablement Services:

```
INVITE sip:05033804036@192.168.135.201;user=phone SIP/2.0
From: 0428387000<sip:0428387000@192.168.135.209;user=phone>;tag=5d8118386-
4de9-097A
To: 05033804036<sip:05033804036@192.168.135.201;user=phone>
Call-ID: 288021b25-003047-1cd7@192.168.135.209
CSeq: 1 INVITE
Via: SIP/2.0/UDP 192.168.135.209:5060;branch=z9hG4bK-7B26-0560c7E8F-015135
Privacy: none
Max-Forwards: 70
Supported: 100rel,timer,replaces
Allow: INVITE,ACK,BYE,CANCEL,PRACK,INFO
Contact: <sip:0428387000@192.168.135.209;user=phone>
Session-Expires: 180
Content-Type: application/sdp
Content-Length:177

v=0
o=hitachi 672204563 672204563 IN IP4 192.168.135.209
s=Session SDP
c=IN IP4 192.168.135.209
t=0 0
m=audio 6004 RTP/AVP 0
a=ptime:20
a=rtpmap:0 PCMU/8000
a=sendrecv
```

Sample SIP INVITE Message from Avaya SIP Enablement Services to NT-SG:

```
INVITE sip:905033804028@192.168.135.209;transport=udp SIP/2.0
Call-ID: 80a4751d627db12a1044a4fb2000
CSeq: 1 INVITE
From: "Aomori 4620SW, "
<sip:05033804036@alj.apac.avaya.com:5061>;tag=80a4751d627db1291044a4fb2000
Record-Route:
<sip:192.168.135.201:5060;lr>,<sip:192.168.135.141:5061;lr;transport=tls>
To: "905033804028" <sip:905033804028@alj.apac.avaya.com>
Via: SIP/2.0/UDP
192.168.135.201:5060;branch=z9hG4bK83838303030316161616479e.0,SIP/2.0/TLS
192.168.135.141;psrrposn=2;branch=z9hG4bK80a4751d627db12b1044a4fb2000
Content-Length: 162
Content-Type: application/sdp
Contact: "Aomori 4620SW, "
<sip:05033804036@192.168.135.141:5061;transport=tls>
Max-Forwards: 69
User-Agent: Avaya CM/R013x.01.1.628.7
Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
Session-Expires: 240;refresher=uac
Min-SE: 240
History-Info: <sip:905033804028@alj.apac.avaya.com>;index=1
History-Info: "905033804028" <sip:905033804028@alj.apac.avaya.com>;index=1.1
Supported: 100rel,timer,replaces,join,histinfo
P-Asserted-Identity:"Aomori 4620SW,
"<sip:05033804036@alj.apac.avaya.com:5061>

v=0
o=- 1 1 IN IP4 192.168.135.141
s=-
c=IN IP4 192.168.135.142
t=0 0
m=audio 4364 RTP/AVP 0 127
a=rtpmap:0 PCMU/8000
a=rtpmap:127 telephone-event/8000
```

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.