Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Carrier Access Adit 3104 IP Business Gateway with Avaya SIP Enablement Services and Avaya Communication Manager - Issue 1.0

Abstract

These Application Notes describe the procedure for configuring the Carrier Access Adit 3104 IP Business Gateway to interoperate with Avaya SIP Enablement Services and Avaya Communication Manager using the Session Initiation Protocol (SIP).

The Carrier Access Adit 3104 IP Business Gateway is an analog to SIP gateway integrated with a 4-port Ethernet switch that has router and firewall capabilities. The Adit 3104 supports Network Address Translation, a DHCP server and a SIP Application Level Gateway. However, the emphasis of the testing was placed on SIP interoperability.

Information in these Application Notes has been obtained through Developer*Connection* compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the procedure for configuring the Carrier Access Adit 3104 IP Business Gateway to interoperate with Avaya SIP Enablement Services and Avaya Communication Manager using the Session Initiation Protocol (SIP).

The Carrier Access Adit 3104 IP Business Gateway is an analog to SIP gateway integrated with a 4-port Ethernet switch that has router and firewall capabilities. The Adit 3104 supports Network Address Translation, a DHCP server and a SIP Application Level Gateway.

The Adit 3104 registers with the Avaya SES as a SIP endpoint for each analog telephone connected to the Adit 3104. When a call is placed from an analog telephone, the Adit 3104 will send SIP signaling messages to the Avaya SES to setup the call. Once the call has been setup, the Adit 3104 converts the analog signal from the analog telephone to a series of voice samples sent in data packets over the data network using the Real Time Protocol (RTP).

1.1. Configuration

Figure 1 illustrates the configuration used in these Application Notes. In the sample configuration, two sites are connected via an IP network. The main site has an Avaya SES and Avaya S8300 Media Server running Avaya Communication Manager in an Avaya G350 Media Gateway. Endpoints include an Avaya 4600 Series IP Telephone (with SIP firmware), an Avaya 4600 Series IP Telephone (with H.323 firmware) and an Avaya 6408D Digital Telephone. An ISDN-PRI trunk connects the media gateway to the Public Switched Telephone Network (PSTN). Avaya IA770 Intuity AUDIX is installed on the Avaya S8300 Media Server for voicemail.

The branch site has a Carrier Access Adit 3104 IP Business Gateway with two analog telephones connected to it. The branch site also has two Avaya 4600 Series IP Telephones (with SIP firmware) and a Windows PC connected to the Ethernet switch ports of the Adit 3104. A Windows PC running an Internet browser is connected to the public network for management of the Adit 3104. This PC also serves as a TFTP server for the Avaya IP Telephones at the branch site.

All SIP telephones and analog telephones at both sites are registered to Avaya SES and are administered as Outboard Proxy SIP (OPS) stations in Avaya Communication Manager. However, the SIP telephones at the branch site are configured to use the Adit 3104 IP address as the default gateway. Thus, all SIP traffic between the endpoints and the Avaya SES will pass through the Adit 3104. The Adit 3104 is configured to be the DHCP server for the branch site. It will provide the IP addresses for the PC and SIP telephones.

The two DID numbers of the ISDN-PRI trunk to the Main Site are each mapped to a telephone extension at the Main Site.

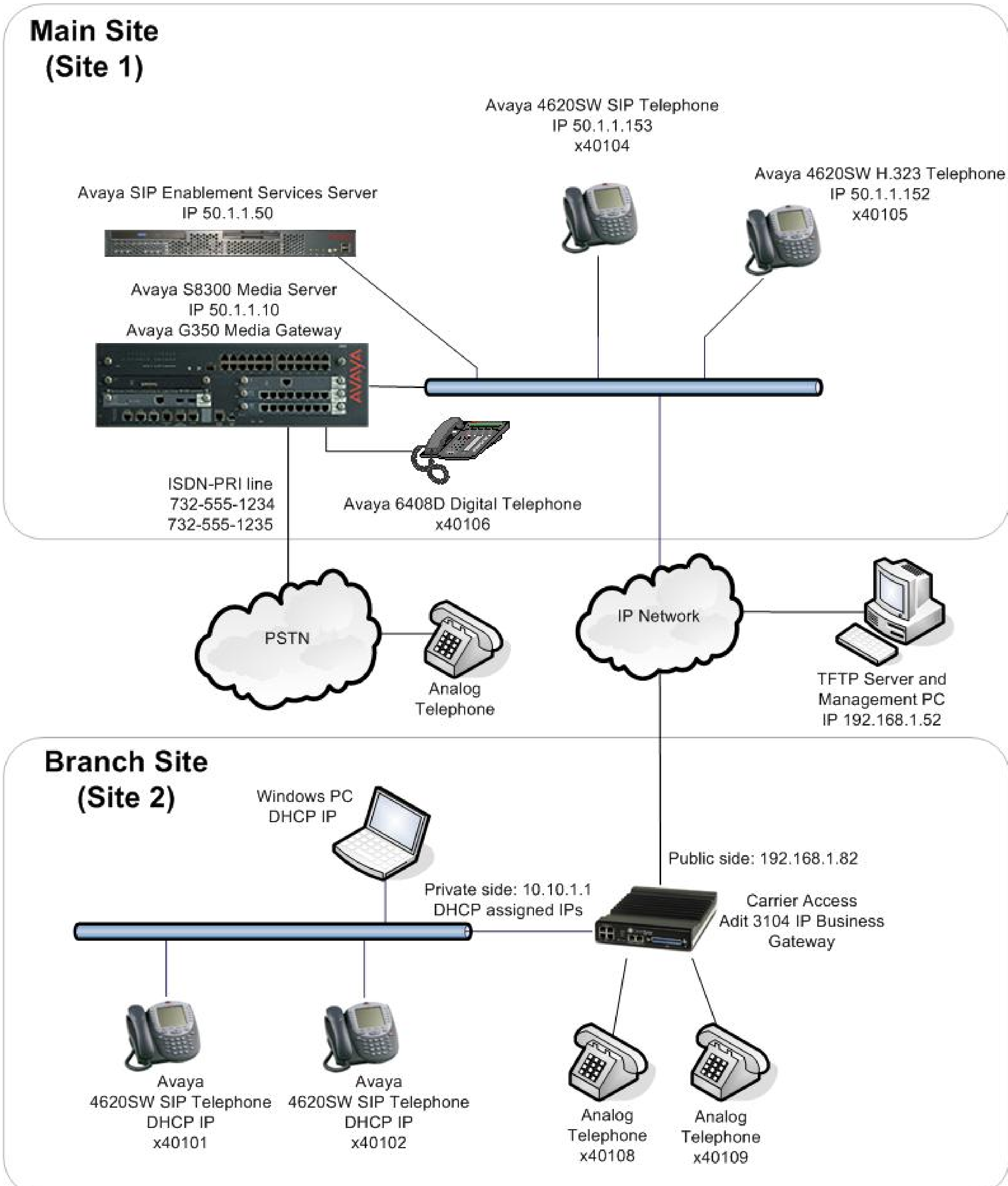


Figure 1: Adit 3104 Test Configuration

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8300 Media Server with Avaya G350 Media Gateway. Avaya IA770 Intuity Audix is included in the installation.	Avaya Communication Manager 3.1.2 (R013x.01.2.632.1) with Service Pack (01.2.632.1-11989)
Avaya SIP Enablement Services (SES)	3.1 (build 18)
Avaya 4620SW IP Telephones	SIP version 2.2.2 H.323 version 2.3
Avaya 6408D Digital Telephone	-
Analog Telephones	-
Windows PCs	Windows XP Professional
Carrier Access Adit 3104 IP Business Gateway	1.4.0.26

3. Configure Avaya Communication Manager

The communication between Avaya Communication Manager and Avaya SES is via a SIP trunk group. All SIP signaling for calls between Avaya Communication Manager and the Adit 3104 passes through Avaya SES via this trunk group. This section describes the steps for configuring this trunk group, and associated signaling group. In addition, this section describes the configuration of stations as OPS stations, which is required for each analog telephone connected to the Adit 3104.

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

Step	Description
1.	Use the **display system-parameters customer-options** command to verify that sufficient SIP trunk capacity exists. On Page 2, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. Thus, a call from a SIP telephone to another SIP telephone will use two SIP trunks. A call between a non-SIP telephone and a SIP telephone will only use one trunk. In this solution, each analog endpoint at the branch counts as a SIP telephone. 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.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 100   10
           Maximum Concurrently Registered IP Stations: 20    0
             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: 5     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: 10    0

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

Step	Description
2.	Use the **change node-name ip** command to assign the node name and IP address for Avaya SES at the enterprise site. In this case, ***SES*** and ***50.1.1.50*** 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 ***50.1.1.10*** 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                 50 .1   .1   .50                           .   .   .
default             0  .0   .0   .0                            .   .   .
procr               50 .1   .1   .10                           .   .   .
```

Step	Description
3.	Use the **change ip-network-region *n*** command, where ***n*** is the number of the region to be changed, to define the connectivity settings for all VoIP resources and IP endpoints within the 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 Step 5. In the case of the compliance test, the same IP network region that contains the Avaya S8300 Media Server and Avaya IP Telephones was selected to contain the Avaya SES server. By default, the Media Server and IP telephones are in IP Network Region 1. On 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. If different IP network regions are used for the Avaya S8300 Media Server and the Avaya SES server, 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: 1       Authoritative Domain: devcon.com
    Name:
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: 3027
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 34      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      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
```

Step	Description
4.	Use the **change ip-codec-set *n*** command, where ***n*** is the codec set value specified in Step 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. The list should include the codecs the enterprise wishes to support within the normal trade-off of bandwidth versus voice quality. The example below shows the values used in the compliance test. ``` change ip-codec-set 1 Page 1 of 2 IP Codec Set Codec Set: 1 Audio Silence Frames Packet Codec Suppression Per Pkt Size(ms) 1: G.711MU n 2 20 2: G.729AB n 2 20 3: ```

Step	Description
5.	Use the **add signaling group *n*** command, where ***n*** is the number of an unused signaling group, to create the SIP signaling group 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. ▪ 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** and the **Far-end Node Name** fields, respectively. These field values are taken from the **IP Node Names** form shown in Step 2. For alternative configurations that use a C-LAN board, the near (local) end of the SIP signaling group will be the 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 in Step 3. This defines which IP network region contains the Avaya SES server. ▪ 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 is normally set to ***y,*** so that media is sent directly between the endpoints and does not use resources on the Avaya Communication Manager. However, due to an incompatibility between the Adit 3104 and the Avaya SIP Telephones, this field was set to ***n*** for the compliance test. Otherwise, some conferencing scenarios do not succeed between these two types of endpoints. ▪ 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? n` ` Session Establishment Timer(min): 120`

Step	Description
6.	Add a SIP trunk group by using the **add trunk-group *n*** command, where ***n*** is the number of an unused trunk group. For the compliance test, trunk group number 1 was chosen. On Page 1, set the fields to the following values: ▪ Set the **Group Type** field to ***sip***. ▪ Choose a descriptive **Group Name**. ▪ Specify an available trunk access code (**TAC**) that is consistent with the existing 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 Step 5. ▪ Specify the **Number of Members** supported by this SIP trunk group. As mentioned earlier, each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. Thus, a call from a SIP telephone to another SIP telephone will use two SIP trunks. A call between a non-SIP telephone and a SIP telephone will only use one trunk. In this solution, each analog endpoint at the branch counts as a SIP telephone. ▪ The default values may be retained for the other fields.

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

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: To SES 50.1.1.50          COR: 1        TN: 1        TAC: 101
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                        Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                      Signaling Group: 1
                                                    Number of Members: 24
```

Step	Description
7.	On Page 3: ▪ Verify the **Numbering Format** field is set to ***public***. This field specifies the format of the calling party number sent to the far-end. ▪ The default values may be retained for the other fields.

```
add trunk-group 1                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y



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


                                                 Replace Unavailable Numbers? n
```

Step	Description
8.	Use the **change public-unknown numbering 0** command to define the full calling party and connected party number to be sent to the far-end. Add an entry for the trunk group defined in Step 6. In the example shown below, all calls originating from a 5 digit extension beginning with 4 and routed across trunk group 1 will be sent as a 5 digit calling number. 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 4 1 5 ```
9.	Create a route pattern that will use the SIP trunk that connects to Avaya SES. In general, a route pattern is not required for calling between SIP endpoints registered to the Avaya SES. This includes the dialing scenarios performed in the compliance test. However, some transfer scenarios using alpha-numeric handles (i.e., user names) instead of extensions require a default route pattern. The creation of this default route pattern is included here for completeness. To create a route pattern, use the **change route-pattern *n*** command, where ***n*** is the number of an unused route pattern. Enter a descriptive name for the **Pattern Name** field. Set the **Grp No** field to the trunk group number created for the SIP trunk. Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of ***0*** is the least restrictive level. The default values may be retained for all other fields. ``` change route-pattern 1 Page 1 of 3 Pattern Number: 3 Pattern Name: SIP 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 n user 2: n user 3: n user 4: n user 5: n user 6: n user BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM No. Numbering LAR 0 1 2 3 4 W Request Dgts Format Subaddress 1: y y y y y n n rest none 2: y y y y y n n rest none 3: y y y y y n n rest none 4: y y y y y n n rest none 5: y y y y y n n rest none 6: y y y y y n n rest none ```

Step	Description
10.	Use the **change locations** command to assign the default SIP route pattern to the location. In the compliance test, all SIP endpoints whether at the main or branch site are part of a single location defined in Avaya Communication Manager. This location uses the default name of ***Main*** and is shown in the example below. The **Name** field can be changed to any descriptive name. Enter the route pattern number from the previous step in the **Proxy Sel. Rte. Pat.** field. The default values may be retained 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:`
11.	All SIP stations are configured as OPS stations on Avaya Communication Manager. This includes the analog telephones connected to the Adit 3104 which appear as SIP stations to Avaya Communication Manager. Use the **display system-parameters customer-options** command to verify Avaya Communication Manager has sufficient OPS capacity available to add the OPS stations needed for the SIP and analog telephones at the branch office in **Figure 1**. If there is insufficient capacity, contact an authorized Avaya sales representative or business partner to make the appropriate changes. `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   121` `                                     Maximum Stations: 450   41` `                             Maximum XMOBILE Stations: 0     0` `                   Maximum Off-PBX Telephones - EC500: 50    0` `                   Maximum Off-PBX Telephones -   OPS: 50    23` `                   Maximum Off-PBX Telephones - SCCAN: 0     0`

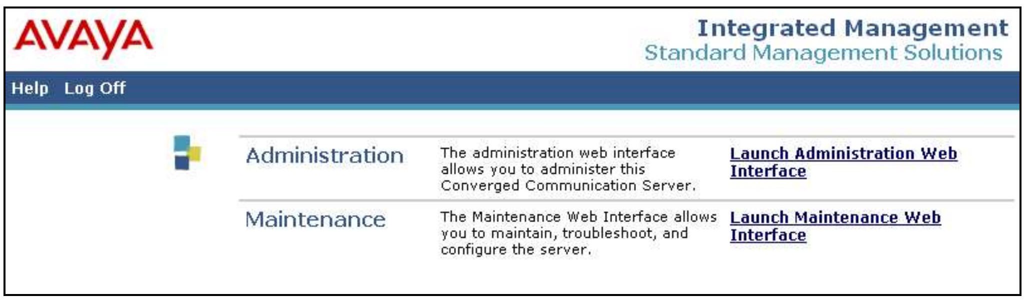
Step	Description
12.	To add a station, use the **add station *n*** command where ***n*** is an unused extension number. Use the default value of ***6408D+*** for the **Type** field. Enter an ***X*** in the **Port** field. This indicates a station is being added without identifying a physical port for the station to use. Enter a descriptive name in the **Name** field. The **Coverage Path 1** field is set to ***1***. Coverage path 1 directs the call to voicemail. The voicemail configuration is not covered in these Application Notes. The default values may be retained for all other fields. <pre>add station 40108 Page 1 of 4 STATION Extension: 40108 Lock Messages? n BCC: 0 Type: 6408D+ Security Code: TN: 1 Port: X Coverage Path 1: 1 COR: 1 Name: Branch 1 Coverage Path 2: COS: 1 Hunt-to Station: STATION OPTIONS Loss Group: 2 Personalized Ringing Pattern: 1 Data Module? n Message Lamp Ext: 40108 Speakerphone: 2-way Mute Button Enabled? y Display Language: english Media Complex Ext: IP SoftPhone? n</pre>
13.	On Page 2, set **Restrict Last Appearance** to ***n***. This will allow the last call appearance to be used for either an incoming or outgoing call. <pre>add station 40108 Page 2 of 5 STATION FEATURE OPTIONS LWC Reception: audix 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? y Restrict Last Appearance? n Active Station Ringing: single H.320 Conversion? n Per Station CPN - Send Calling Number? Service Link Mode: as-needed Multimedia Mode: basic Audible Message Waiting? n MWI Served User Type: Display Client Redirection? n AUDIX Name: IA770 Select Last Used Appearance? n Coverage After Forwarding? s Direct IP-IP Audio Connections? y Emergency Location Ext: 40108 IP Audio Hairpinning? n</pre>

Step	Description
14.	On Page 3, under **BUTTON ASSIGNMENTS**, create the appropriate number of call appearances for the SIP endpoint being configured. In general, the appropriate number of call appearances on Avaya Communication Manager is the same as the number of call appearances supported by the endpoint. To create a call appearance, enter ***call-appr*** as the button assignment. The example below shows the configuration of one of the analog phones connected to the Adit 3104. The analog phones that were used, supported two call appearances. `add station 40108                                           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:                                    7:` ` 4:                                    8:`
15.	Map the Avaya Communication Manager extension to the Avaya SES media server extension defined in Section 4, Step 8 with the **add off-pbx-telephone station-mapping** command. Enter the values as shown below: ▪ **Station Extension**: Avaya Communication Manager extension created in Step 12. ▪ **Application**: ***OPS*** ▪ **Phone Number**: Avaya SES media server extension ▪ **Trunk Selection**: The SIP trunk group number ▪ **Configuration Set**: Enter a valid configuration set. The compliance test used configuration set 1 which contained the default values. `add off-pbx-telephone station-mapping                       Page   1 of   2` `                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION` ` Station     Application   Dial  Phone Number      Trunk       Configuration` ` Extension                 Prefix                  Selection   Set` ` 40108         OPS              - 40108             1          1` `                                 -`

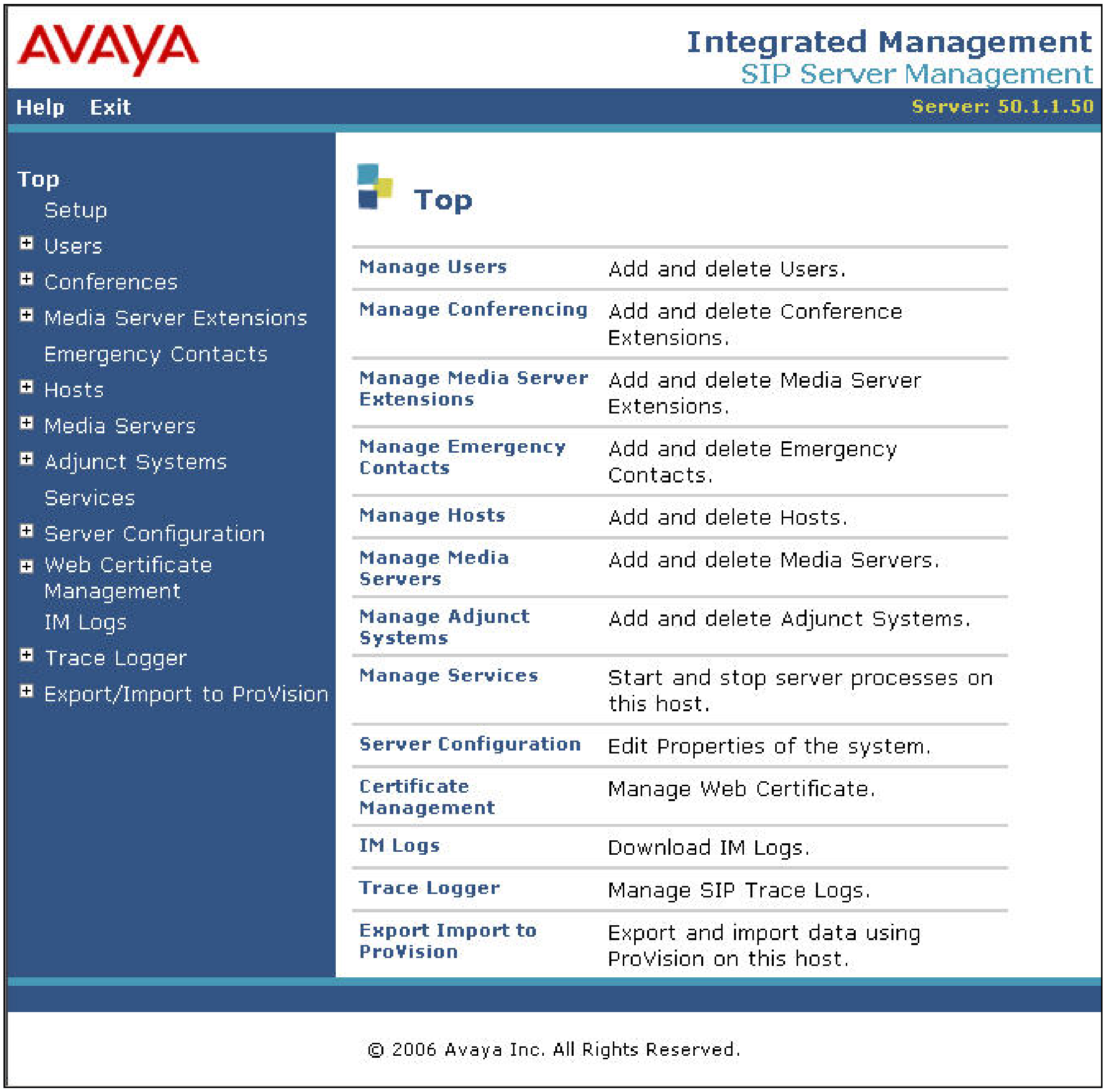
Step	Description
16.	On Page 2, set the **Call Limit** to the number of call appearances set on the station form in Step 14. Verify that the **Mapping Mode** is set to ***both***. `add off-pbx-telephone station-mapping                          Page   2 of   2` `STATIONS WITH OFF-PBX TELEPHONE INTEGRATION` `Station        Call       Mapping      Calls       Bridged` `Extension      Limit      Mode         Allowed     Calls` `40108          2          both         all         both`
17.	Repeat Steps 12 - 16 for each remaining station located at the branch office. The branch office has four stations: two analog telephones connected to the Adit 3104 (x40108 and x40109) and two Avaya 4600 Series SIP Telephones (x40101 and x40102).
18.	To map a DID number to a station at the main or branch office, use the **change inc-call-handling-trmt trunk-group *n*** command, where ***n*** is the trunk group number connected to the PSTN from the Avaya G350 Media Gateway. The compliance test used trunk group 2 to connect to the PSTN. This trunk group configuration is not shown in these Application Notes. The example below shows two incoming 11-digit numbers being deleted and replaced with the extension number of the desired station. `change inc-call-handling-trmt trunk-group 2                    Page   1 of   3` `INCOMING CALL HANDLING TREATMENT` `Service/       Called    Called        Del  Insert` `Feature         Len      Number` `tie              11  17325551234       11   40104` `tie              11  17325551235       11   40108`

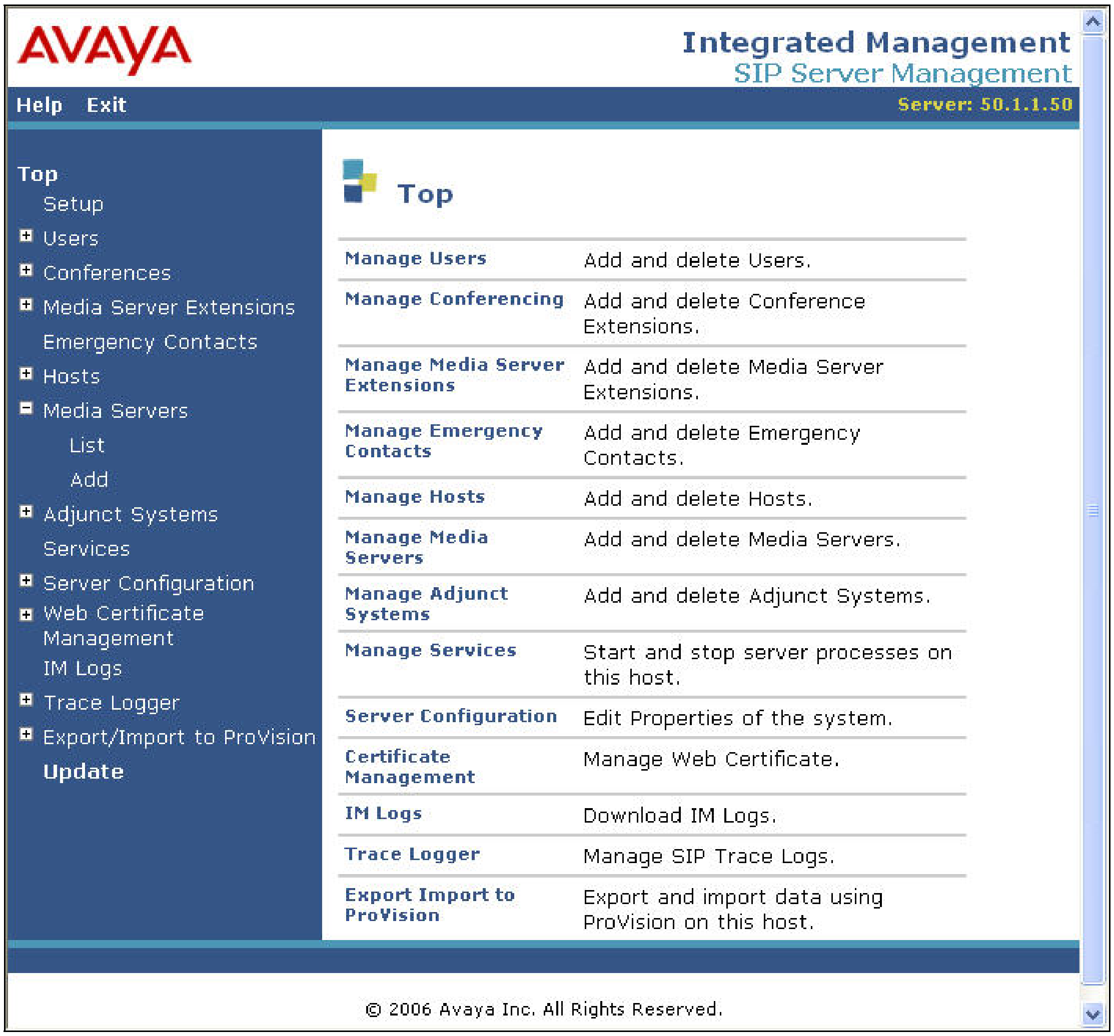
4. Configure Avaya SES

This section covers the configuration 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 from 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].

Step	Description
1.	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 the Avaya SES server. Log in with the appropriate credentials and then select the **Launch Administration Web Interface** link from the main page as shown below. 

Step	Description
2.	The Avaya SES Administration Home Page will be displayed as shown below.



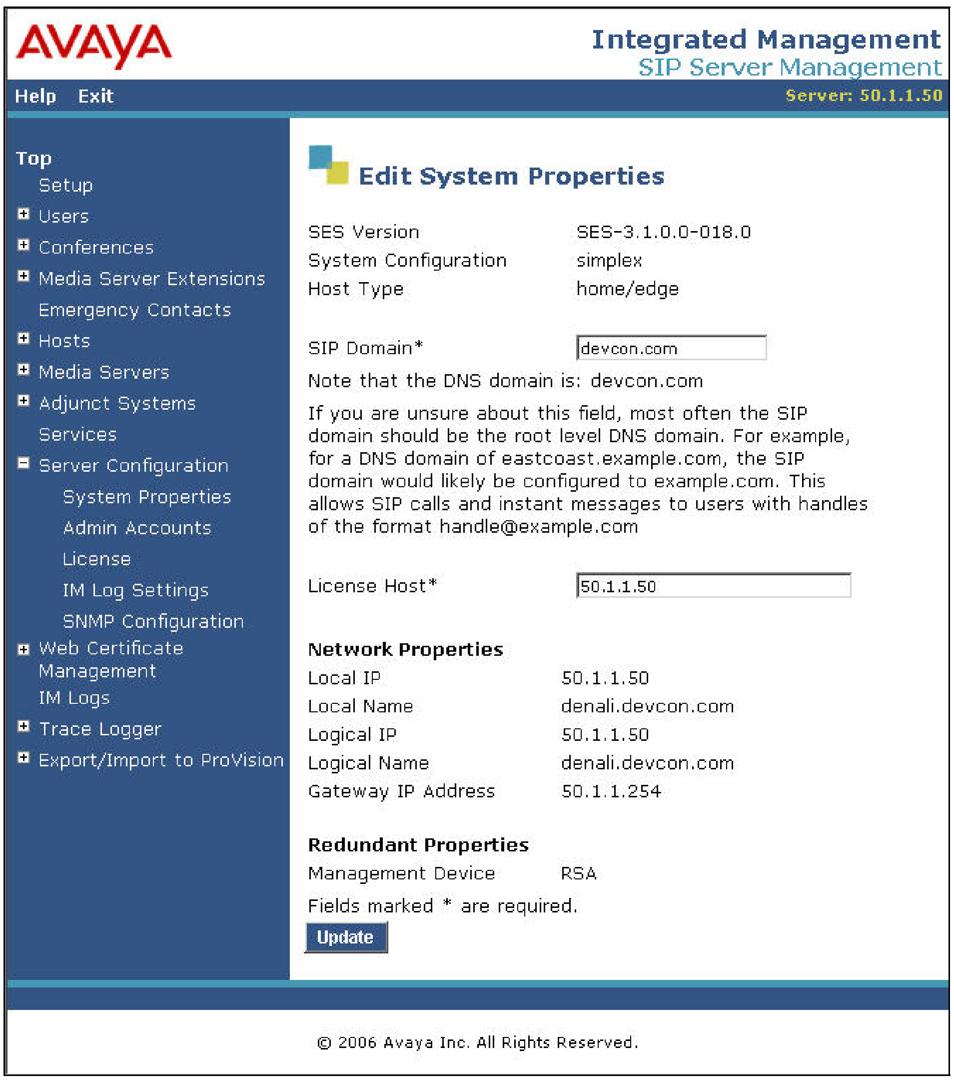
Step	Description
3.	After making changes within Avaya SES, it is necessary to commit the database changes using the **Update** link that appears when changes are pending. Perform this step by clicking on the **Update** link found in the bottom of the blue navigation bar on the left side of any of the Avaya SES administration pages as shown below. It is recommended that this be done after making each set of changes described in the following steps. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>

From the left pane of the administration web interface, expand the **Server Configuration** option and select **System Properties**. The **Edit System Properties** page displays the software version in the **SES Version** field and the network properties entered 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 Section 3, Step 3.
- 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.
- After configuring the **Edit System Properties** page, click the **Update** button.



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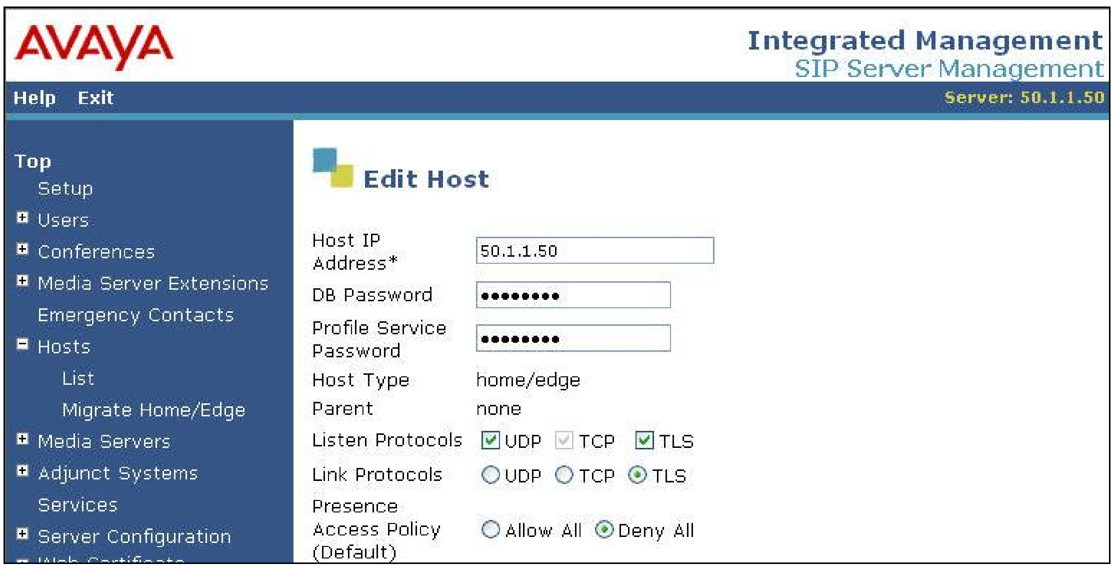
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.</td><td>

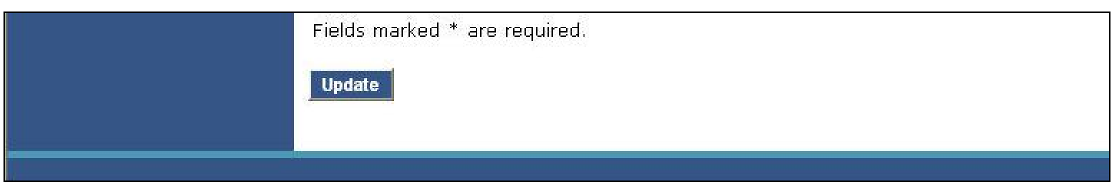
After setting up the domain on the **Edit System Properties** page, create a host computer entry for Avaya SES. The following example shows the **Edit Host** page since the host had already been added to the system.

The **Edit Host** page shown below is accessible by clicking on the **Hosts → List** link in the left pane and then clicking on the **Edit** link under the **Commands** section of the subsequent page that is displayed.

- In the **Host IP Address** field, enter the IP address of the Avaya SES.
- Enter the **DB Password** that was specified during the system installation.
- The default values for the other fields may be used.



- Scroll down to the bottom of the page and click the **Update** button.



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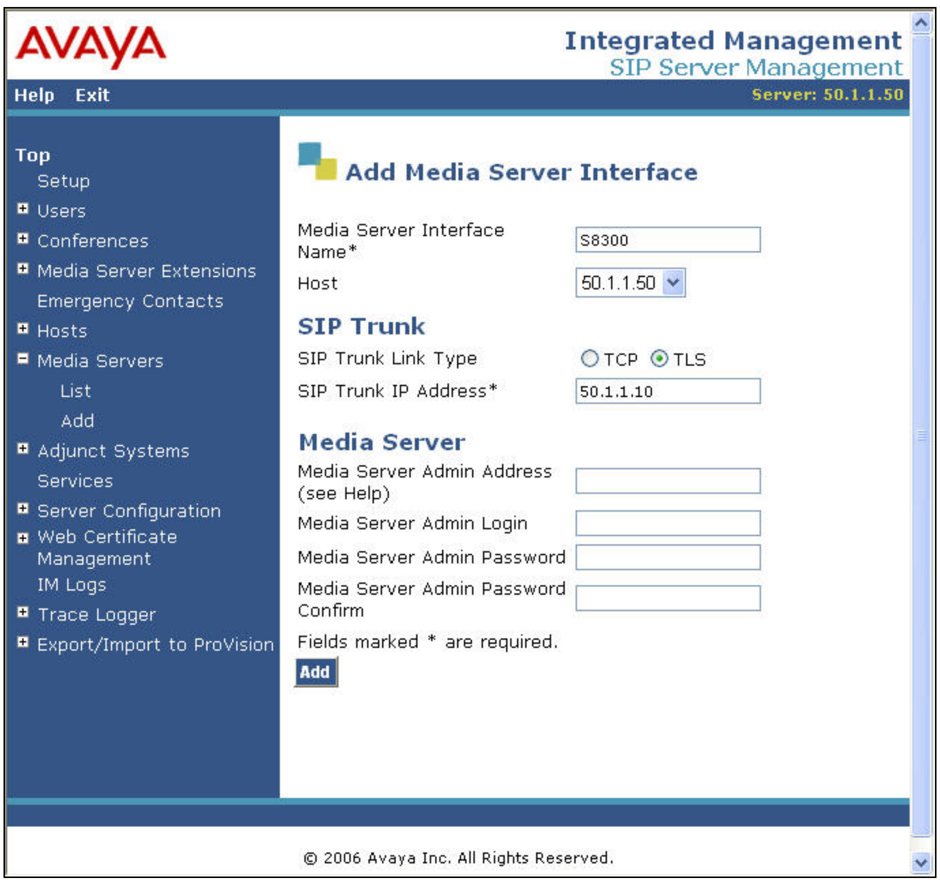
</table>

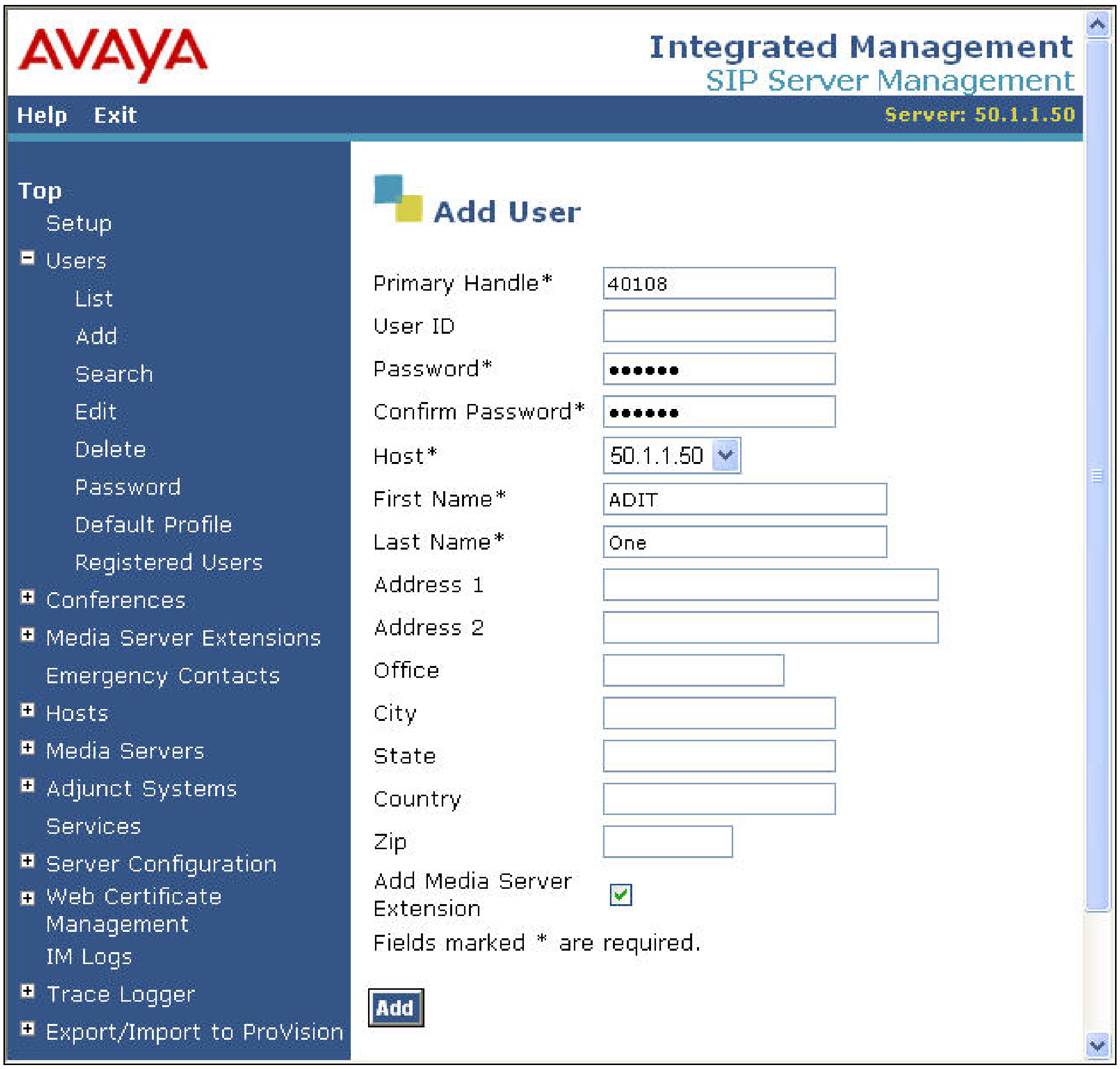
Step	Description
6.	From the left pane of the administration web interface, expand the **Media Servers** option and select **Add** to add the Avaya Media Server to the list of media servers known to Avaya SES. Adding the media server will create the Avaya SES side of the SIP trunk previously created in Avaya Communication Manager. On the **Add Media Server Interface** page, enter the following information: ▪ A descriptive name in the **Media Server Interface Name** field (e.g. S8300). ▪ In the **Host** field, select 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 shown in Step 5. ▪ Select ***TLS*** (Transport Link Security) for the **SIP Trunk 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 alternative configurations that use a C-LAN board, the **SIP Trunk IP Address** would be the IP address of the C-LAN board. ▪ The default values may be retained for all other fields. ▪ After completing the **Add Media Server Interface** page, click the **Add** button. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>7.</td><td>

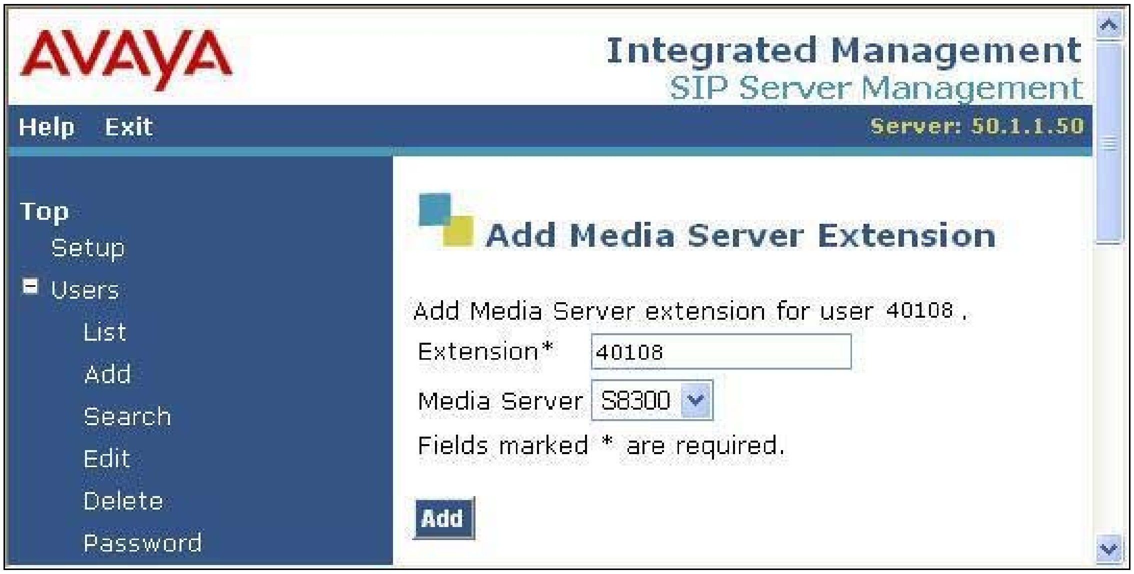
A user must be added on Avaya SES for each of the extensions at the branch office created on Avaya Communication Manager in Section 3, Steps 12 - 16. From the left pane, navigate to **Users** → **Add**. Enter the values as shown below.

- **Primary Handle**: Enter the extension for this user.
- **Password**: Enter a valid password for logging into the SIP endpoint.
- **Confirm Password**: Re-enter the password.
- **Host**: Select the Avaya SES server from the pull-down menu.
- **First Name**: Any descriptive name.
- **Last Name**: Any descriptive name.

Check the **Add Media Server Extension** checkbox. Click the **Add** button to proceed. A confirmation window will appear. Click **Continue** on this new page to proceed.

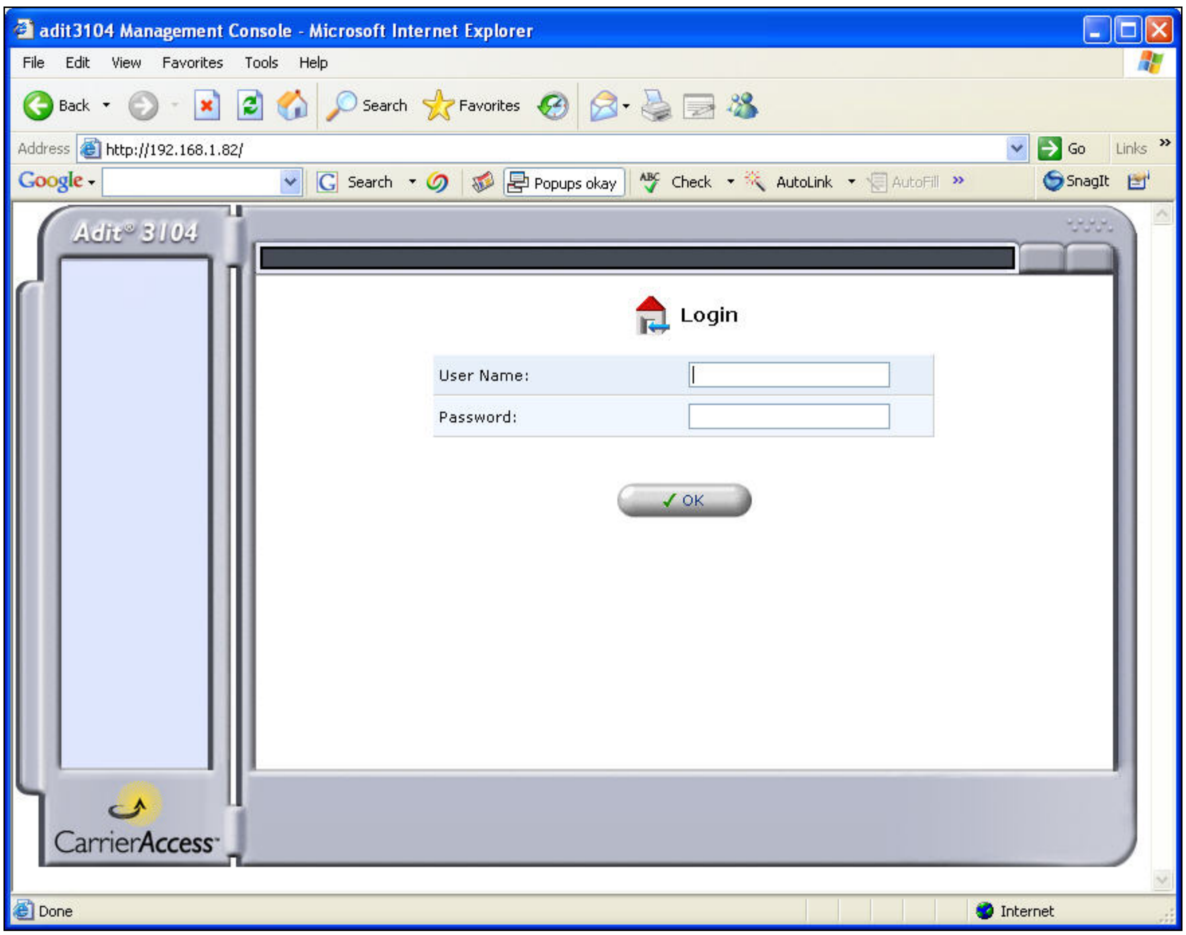


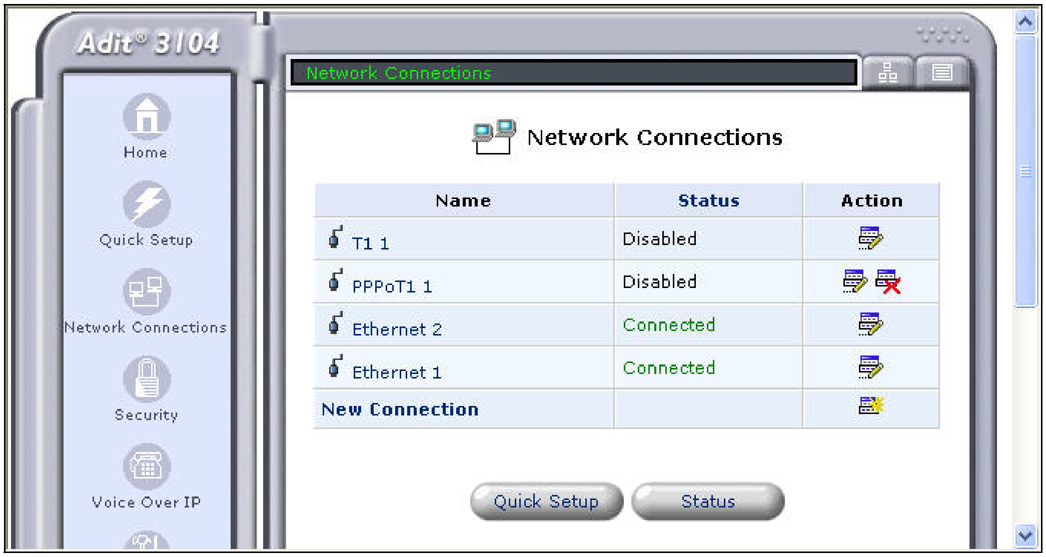
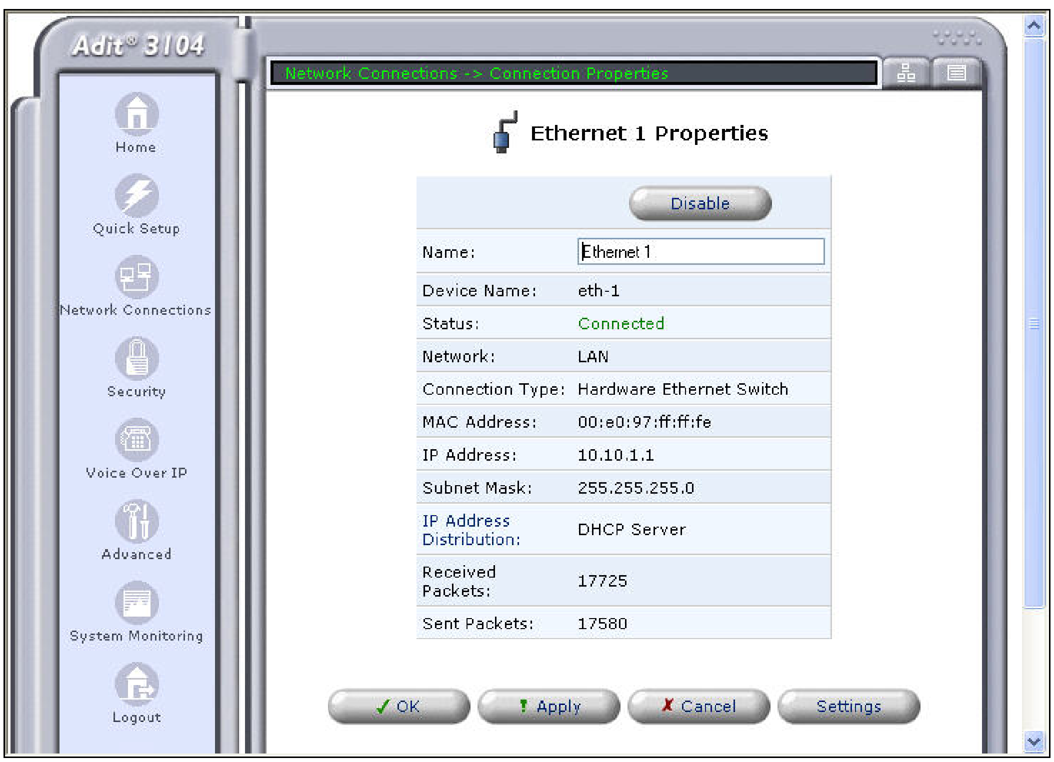
</td></tr>
</table>

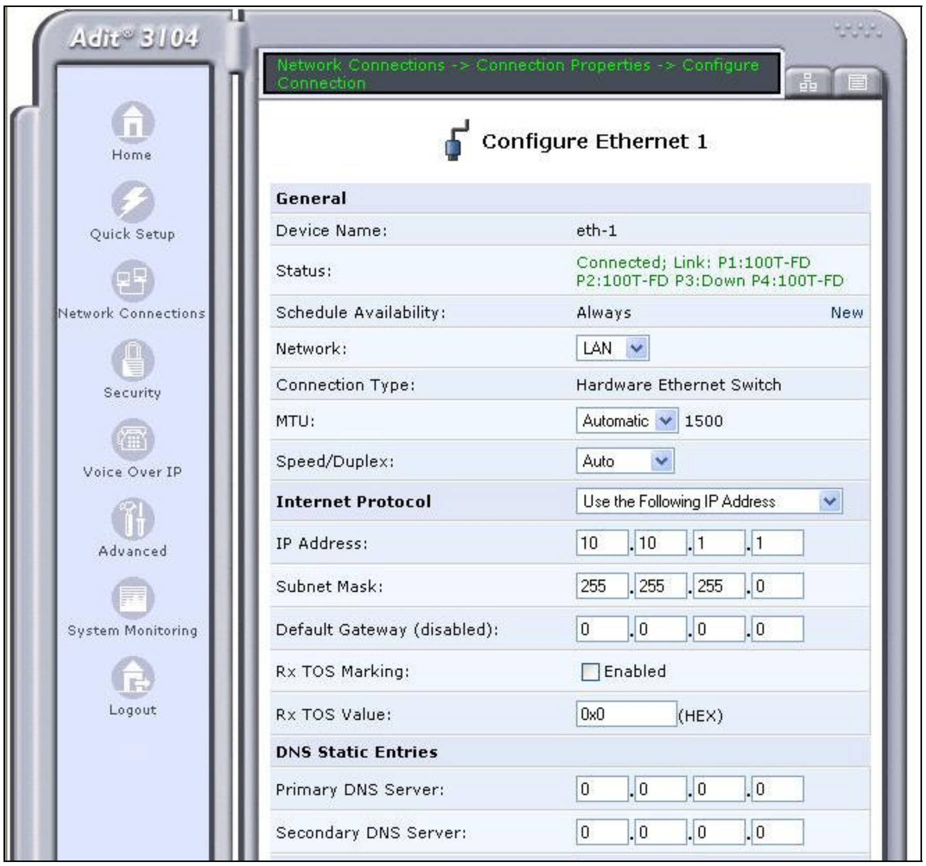
Step	Description
8.	The **Add Media Server Extension** page will appear. In the **Extension** field, enter the same extension used in the previous step. In the **Media Server** field, select from the pull-down menu the name of the media server added in Step 6. Click the **Add** button to complete the operation. 
9.	Repeat Steps 7 - 8 for each of the remaining stations at the branch office.

5. Configure the Adit 3104

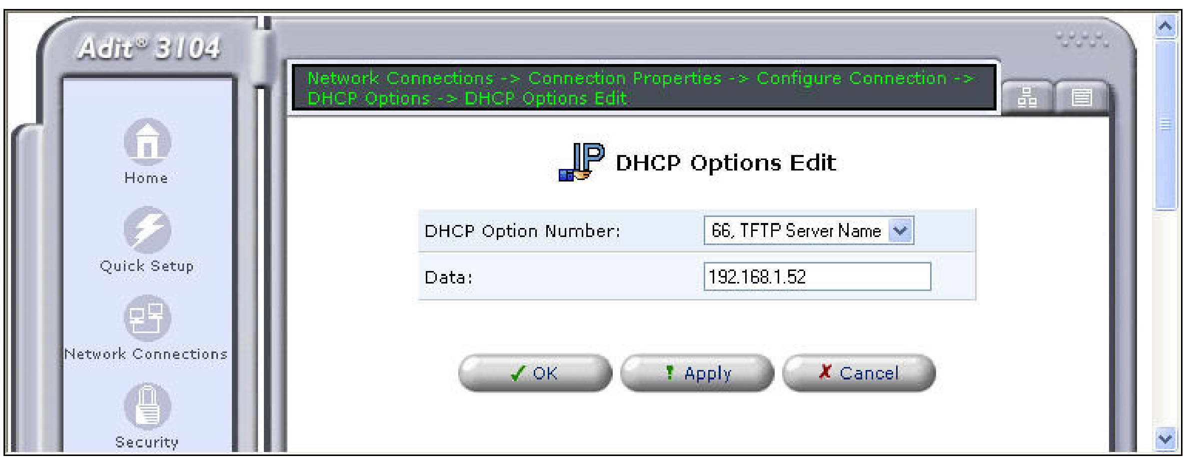
This section describes the procedure for configuring the Adit 3104. This procedure assumes the Adit 3104 has already been configured with IP addresses for both the private and public interfaces. In addition, it is assumed that management access has been enabled on the public interface since the following procedures are performed from a PC on the public side of the device. This is not required for interoperability. The same configuration could be performed from the private side of the device. The Adit 3104 configuration described in this section is performed using an Internet browser. For detailed information on the initial installation of the Adit 3104, consult references [6] and [7].

Step	Description
1.	Launch an Internet browser. Enter the IP address of the Adit 3104 in the **Address** field. The login screen appears as shown below. Enter a valid **User Name** and **Password**. Click **OK** to proceed. 

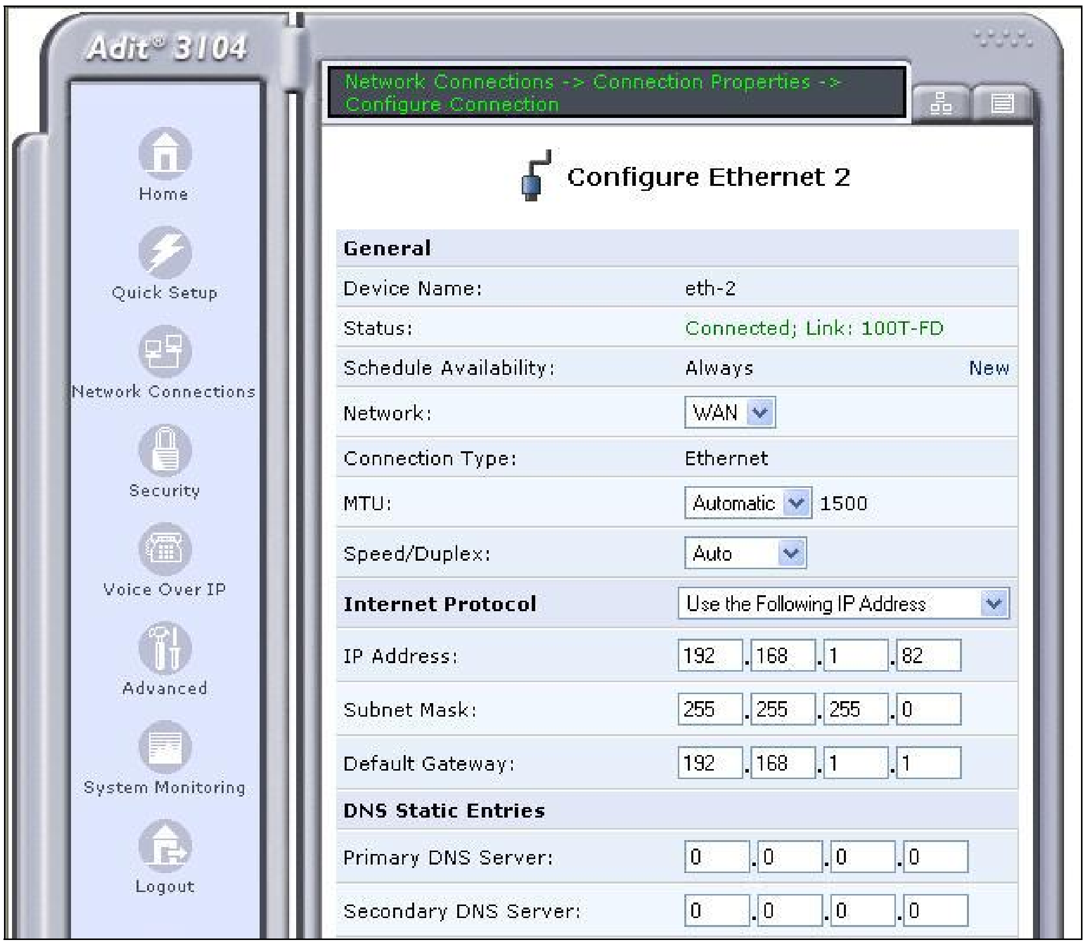
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.</td><td>A list of configuration options will appear in the left pane of the window. To view the properties of the private interface of the device configured during installation, select Network Connections. A list of network connections appears in the right pane. Click the Ethernet 1 entry in the list or the Action icon associated with this entry.

</td></tr>
<tr><td>3.</td><td>A summary of the properties for Ethernet 1 are show in the right pane. Click Settings at the bottom of the pane for the complete list of settings.

</td></tr>
</table>

Step	Description
4.	In the upper half of the right pane, verify the following settings for **Ethernet 1**. Make changes if necessary. ▪ **Network**: Verify ***LAN*** is selected. ▪ **Internet Protocol**: Verify ***Use the Following IP Address*** is selected. ▪ **IP Address:** Verify this field is set to the IP address assigned to the private side of the device. ▪ **Subnet Mask**: Verify the subnet mask is set to an appropriate value for the LAN addressing supported on the private side of the device. ▪ **Default Gateway**: Verify the setting of the default gateway, if one is necessary. In the compliance test, no default gateway is required since the private side LAN is comprised of a single subnet. ▪ The default values may be retained for the other fields. Scroll down to view additional options. 

Step	Description
5.	In the lower half of the right pane, configure the following settings for **Ethernet 1**. ▪ **IP Address Distribution**: Select ***DHCP Server***. This allows the Adit 3104 to serve as a DHCP server for the private LAN side of the device. ▪ **Start IP Address:** Enter the first IP address that can be assigned by the DHCP server. ▪ **End IP Address:** Enter the last IP address that can be assigned by the DHCP server. ▪ **Subnet Mask**: Enter the subnet mask appropriate for the LAN addressing supported on the private side of the device. ▪ The default values may be retained for the other fields. Click the **DHCP Options** link next to the **IP Address Distribution** field to configure the DHCP options. 

Step	Description
6.	A DHCP Option needs to be configured so the DHCP Server can supply the IP address of the TFTP server in the DHCP request. To configure a new DHCP Option, click on the **New Entry** link in the table. 
7.	From the pull-down menu for the **DHCP Option Number** field, select ***66, TFTP Server Name***. In general, Avaya recommends using option 176 to provide this information. However, the Adit 3106 does not support option 176, so option 66 is used instead. In the **Data** field, enter the IP address of the TFTP server. Click **OK**. 

Step	Description
8.	The newly selected option is displayed in the **DHCP Options** table. Click **OK**. The right-pane will return to the **Configure Ethernet** 1 screen. Click OK on this screen to submit any changes. 

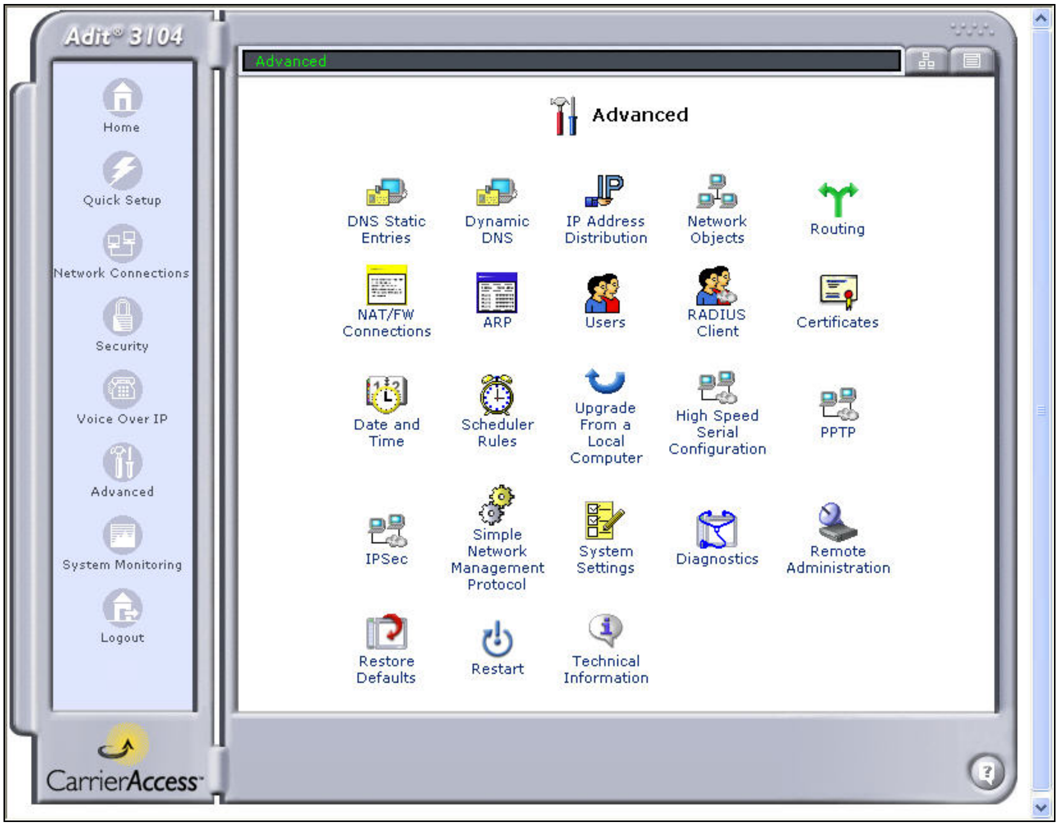
Step	Description
9.	Perform the same procedure described in Steps 2 -3 using **Ethernet 2** to view the properties of the public interface of the device configured during installation. In the upper half of the right pane, verify the following settings for **Ethernet 2**. ▪ **Network:** Verify ***WAN*** is selected. ▪ **Internet Protocol**: Verify ***Use the Following IP Address*** is selected. ▪ **IP Address:** Verify this field is set to the IP address assigned to the public side of the device. ▪ **Subnet Mask**: Verify the subnet mask is an appropriate value for the addressing supported on the public side of the device. ▪ **Default Gateway**: Verify the IP address of the default gateway. ▪ The default values may be retained for the other fields. Scroll down to view additional options. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>10.</td><td>In the lower half of the right pane, configure the following settings for Ethernet 2.

▪ IP Address Distribution: Select Disabled.
▪ Routing Mode: Select NAPT. This enables the Adit 3104 to perform Network Address Translation between the public and private interfaces.
▪ SIP ALG: Check the check box. This enables the Adit 3104 to translate the IP address in the SIP messages between the public and private interfaces.
▪ The default values may be retained for the other fields.

Click OK.

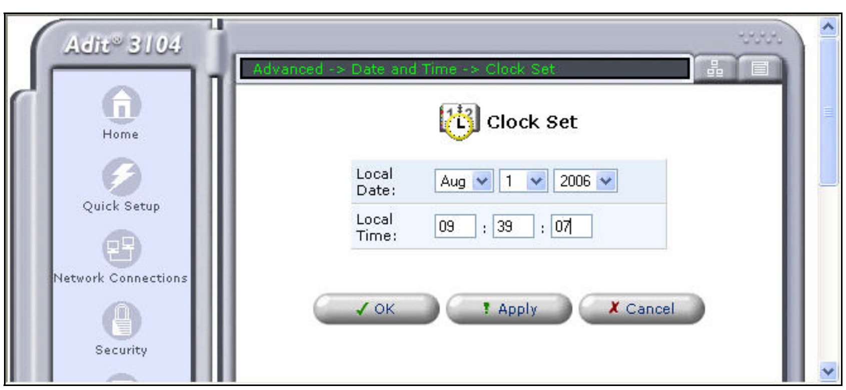

</td></tr>
<tr><td>11.</td><td>Reboot all Avaya SIP telephones at the branch so the telephones will make a DHCP request to the Adit 3104 for an IP address and TFTP server address.</td></tr>
</table>

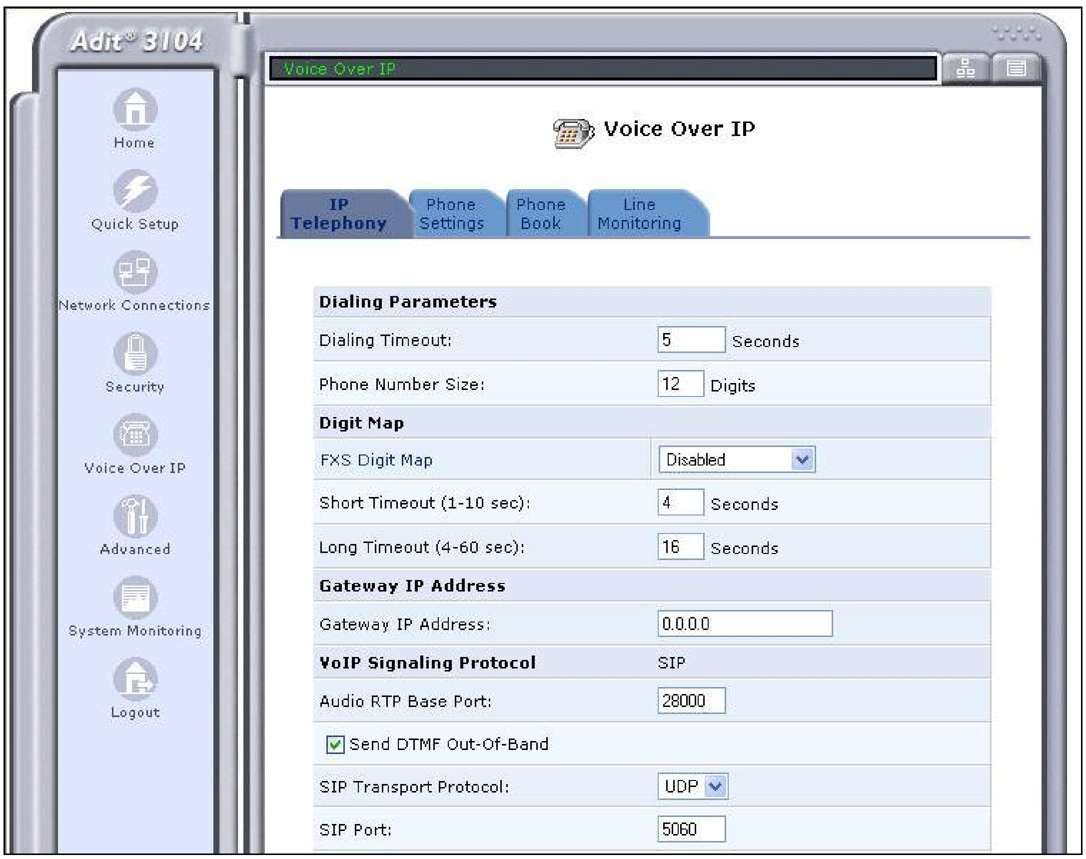
Step	Description
12.	Set the date and time. Navigate to **Advanced → Date and Time**. 

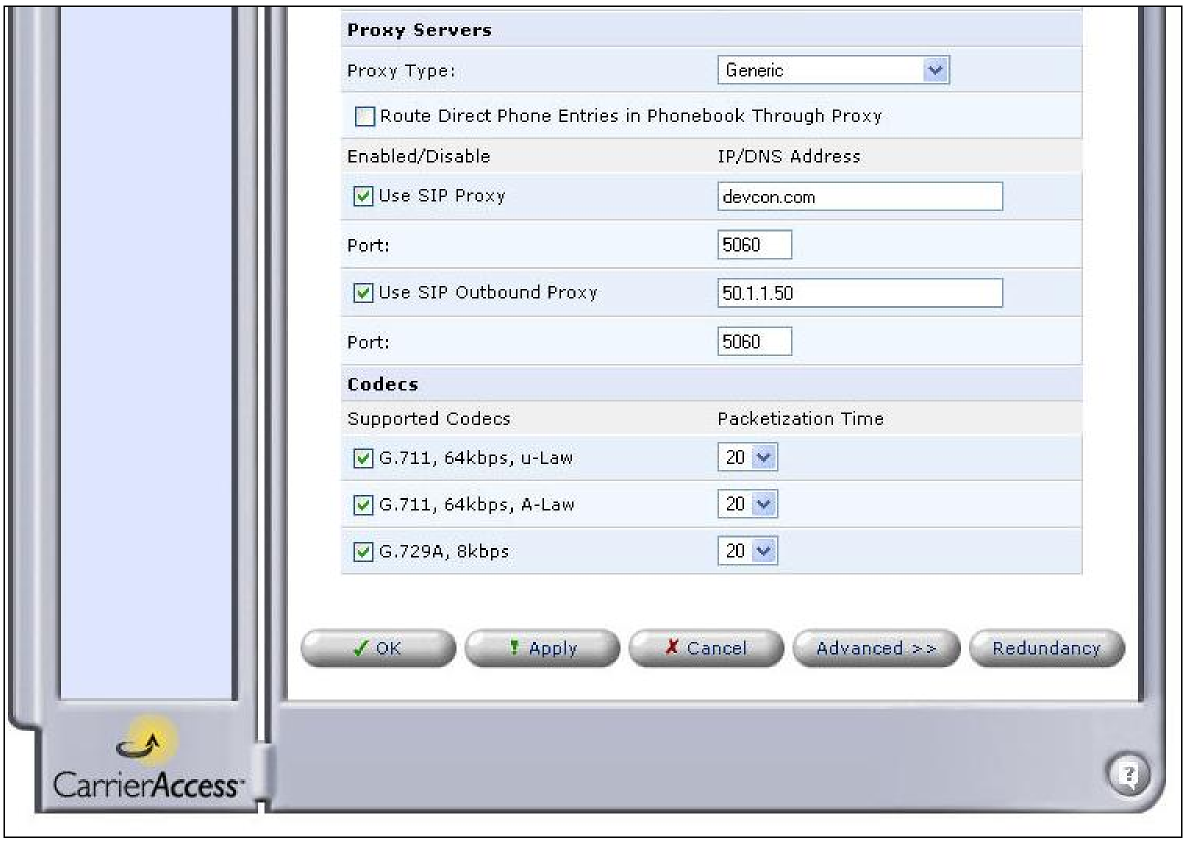
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>13.</td><td>The Data and Time screen appears in the right pane. Click the Clock Set button.

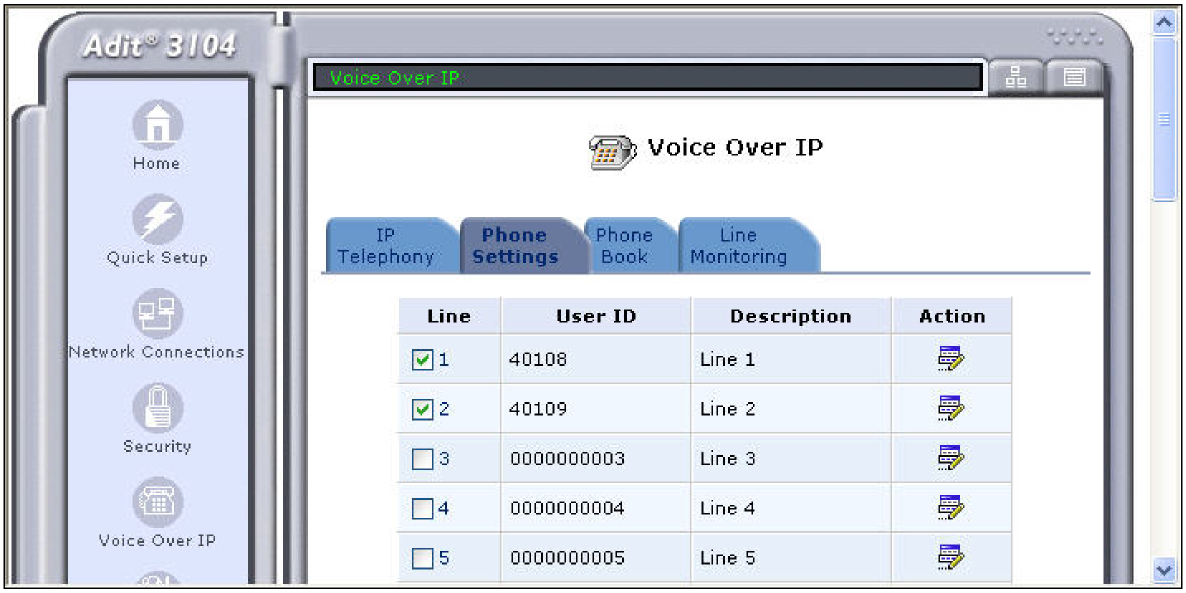

</td></tr>
<tr><td>14.</td><td>In the Clock Set screen, select the Local Date from the pull-down menus. Enter the Local Time. Click OK.

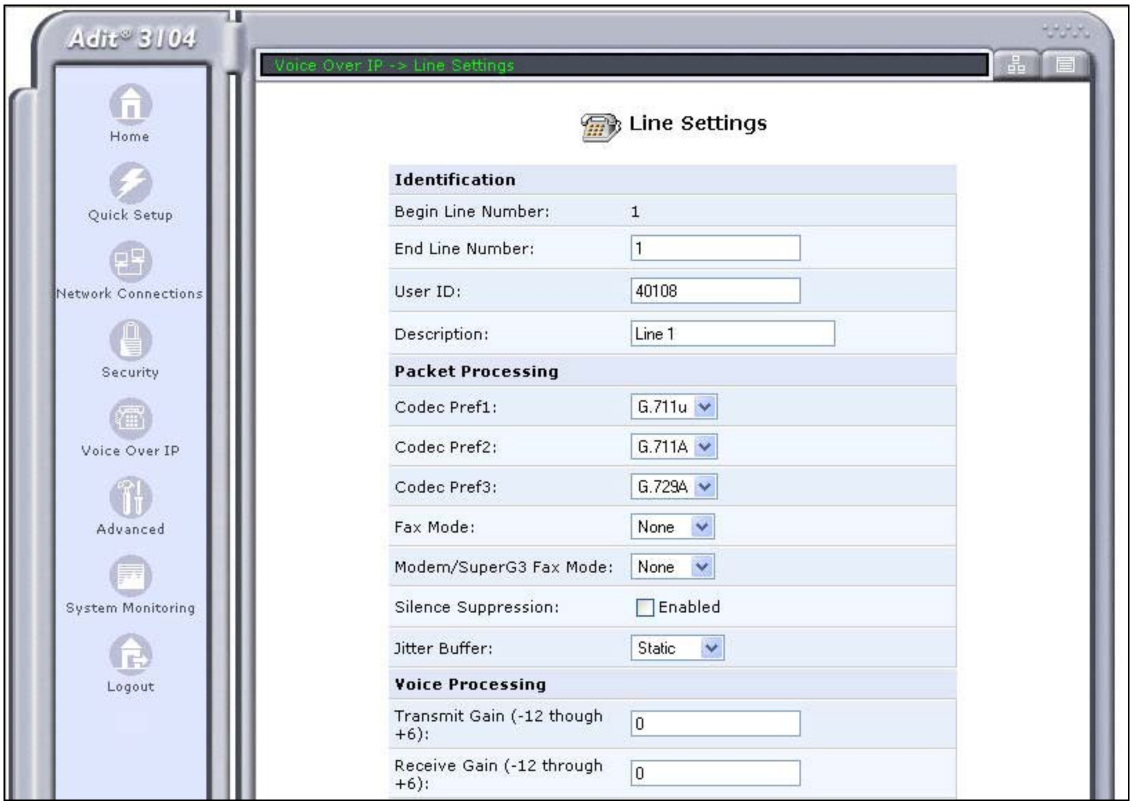
The right-pane will return to the Date and Time screen above. Click OK on this screen also to submit the changes.

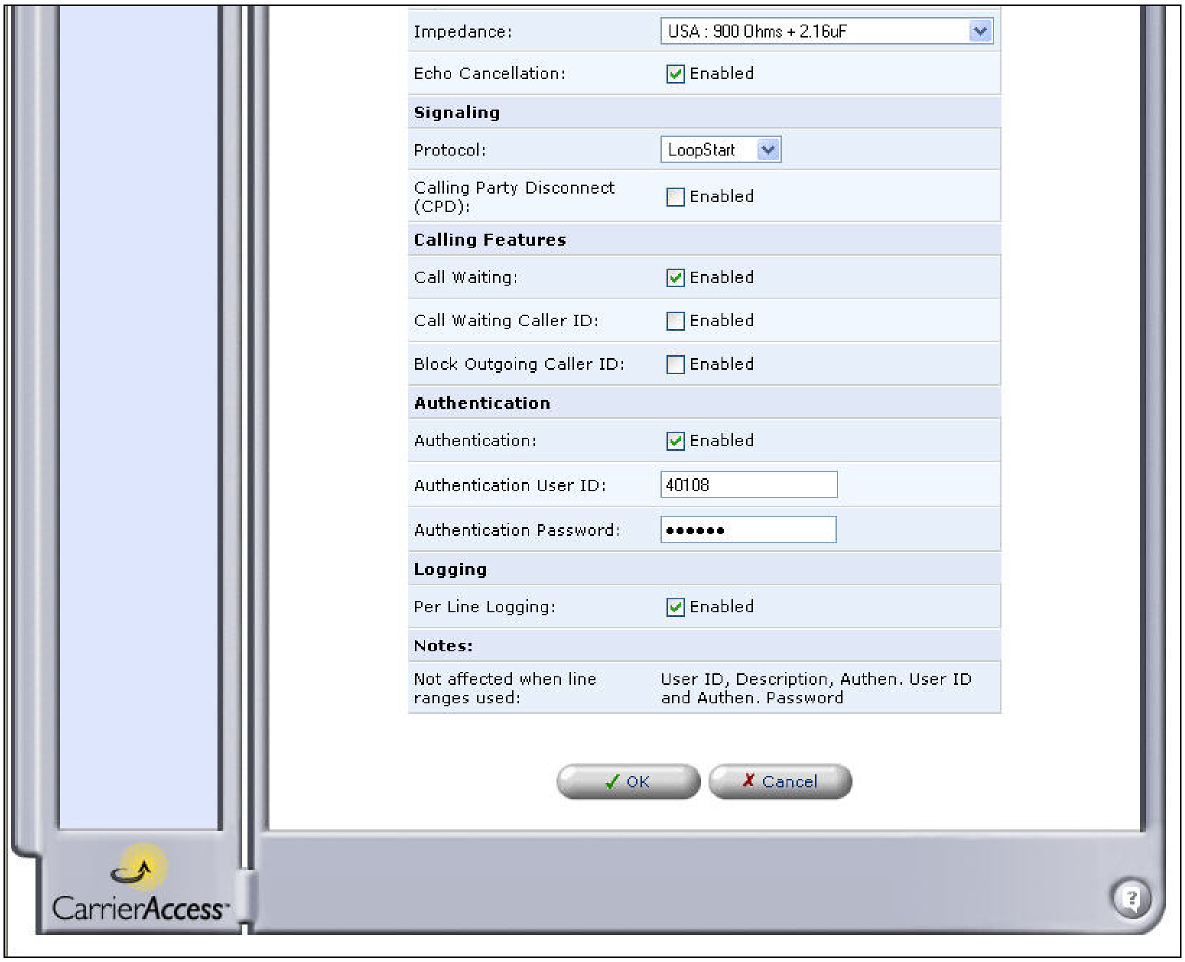

</td></tr>
</table>

Step	Description
15.	To configure the Voice over IP parameters of the Adit 3104, select **Voice Over IP** in the left pane. The **Voice Over IP** screen appears. Select the **IP Telephony** tab in the right pane. In the upper half of the screen, configure or verify the following fields as described below: ▪ **Dialing Timeout**: Enter the maximum time to wait for the user to complete dialing. ▪ **Phone Number Size**: Enter the maximum phone number size. The compliance test used 12 digits to accommodate a 1 digit feature access code plus an 11 digit phone number. ▪ **Send DTMF Out-Of-Band**: Check the check box. ▪ **SIP Transport Protocol**: ***UDP*** ▪ **SIP Port**: ***5060*** ▪ The default values may be retained for the other fields. Scroll down to view additional options. 

Step	Description
16.	In the lower half of the screen, configure or verify the following fields as described below: ▪ **Use SIP Proxy**: Check the check box. Enter the SIP domain configured in the Avaya SES in Section 4, Step 4. ▪ **Port**: ***5060*** ▪ **Use SIP Outbound Proxy**: Check the check box. Enter the IP address of the Avaya SES as showed in Section 4, Step 5. ▪ **Port**: ***5060*** ▪ **Supported Codecs**: For each codec that will be supported by the device, place a check in the check box next to the codec. By default, all are selected. At a minimum, there must be at least one codec selected that is also in the codec list supported on Avaya Communication Manager defined in Section 3, Step 4. ▪ The default values may be retained for the other fields. Click **OK**. 

Step	Description
17.	Return to the main **Voice Over IP** screen. Select the **Phone Settings** tab to configure the properties of each line/port where an analog phone is connected. The screen below shows the **Voice Over IP** screen after the lines have been configured since the **User ID** column show the extensions instead of the default IDs. To activate a line, check the check box next to the line. To configure the line, click the **Line** number or **Action** icon associated with this line in the right most column of the table. 

Step	Description
18.	The **Line Settings** screen appears. In the upper half of the screen, configure or verify the following fields as described below: ▪ **End Line Number**: Enter the same value as the **Begin Line Number** if configuring each line separately. A range of lines can be configured at the same time with similar values by selecting a value larger than the **Begin Line Number**. ▪ **User ID**: Enter the extension to be used by the analog phone. ▪ **Description**: Enter a descriptive name for this line. ▪ **Codec Pref1 – Codec Pref3**: Select from the pull-down menu the codec to be used for each codec preference. **Codec Pref1** is the highest level preference. ▪ **Silence Suppression**: Uncheck the check box next to **Enabled**. ▪ The default values may be retained for the other fields. Scroll down to view additional options. 

Step	Description
19.	In the lower half of the screen, configure or verify the following fields as described below: ▪ **Authentication**: Check the checkbox next to **Enabled**. ▪ **Authentication User ID**: The same value as the **User ID** in the previous step. ▪ **Authentication Password**: The password configured for this user ID on Avaya SES in Section 4, Step 7. ▪ **Per Line Logging**: For the compliance test, logging was enabled by checking the check box next to **Enabled**. ▪ The default values may be retained for the other fields. Click **OK**. A confirmation window will appear. Click **OK** in this window also. 
20.	Repeat Steps 17 – 19 for each line where an analog phone is connected. After configuring all the lines, return to the **Voice Over IP** screen and click **OK**. A confirmation screen will appear, click **OK** on this screen to submit the changes.

6. Interoperability Compliance Testing

This section describes the compliance testing used to verify the interoperability between the Carrier Access Adit 3104 IP Business Gateway, Avaya Communication Manager and Avaya SIP Enablement Services (SES). This section covers the general test approach and the test results.

6.1. General Test Approach

The general test approach was to make calls to/from the telephones connected through the Adit 3104 at the branch site using various codec settings and exercising common PBX features. This testing included the analog telephones and Avaya SIP telephones. The SIP telephones at the branch site register directly with the Avaya SES and use the IP address of the Adit 3104 as the default gateway. The calls were made to/from the main site, the PSTN and within the branch site.

6.2. Test Results

The Adit 3104 successfully passed compliance testing. The following features and functionality were verified during the interoperability compliance test. Each feature was tested with an analog telephone and Avaya SIP telephone, where applicable, unless stated otherwise below:

- Calls to/from the main site
- Calls to/from the PSTN
- Intra-branch calls
- G.711mu and G.729AB codec support
- Proper recognition of DTMF transmissions
- Local device support for Hold, Transfer, and Call Waiting
- Conferencing (SIP phones only)
- Proper system recovery after a Adit 3104 restart
- Proper operation of voicemail with message waiting indicators (MWI). For the analog phones, MWI was only tested using stutter dial tone.
- Extended telephony features using Avaya Communication Manager Feature Name Extensions (FNE) such as Call Forwarding, Call Pickup, Automatic Redial and Send All Calls. For more details on FNEs, please refer to [4].
- NAT'ed PC at the branch was able to connect to external sites.

The following observations were made during the compliance test.

- DHCP Option 176 is not supported. Option 176 is used to provide the TFTP server IP address and other parameters to the Avaya SIP Telephones. However, DHCP Option 66 can be used instead to provide the TFTP server address.
- Some conferencing scenarios are not supported if Direct IP-to-IP Media (shuffling) is enabled. The conference will fail if an Avaya SIP Telephone attempts to conference together two incoming calls; one from another Avaya SIP Telephone and one from an analog telephone connected to the Adit 3104. This is the due to an incompatibility between the Adit 3104 and the Avaya SIP Telephone. Thus, it is recommended that shuffling be disabled on the signaling group of the SIP trunk on Avaya Communication Manager (see Section 3, Step 5). This will prevent the Avaya Communication Manager from attempting to shuffle media between any SIP endpoints.
- When using a codec setting of G.729A without silence suppression in the Adit 3104, the Adit 3104 does not send the parameter annexb=no (silence suppression disabled) in the SIP SDP information. The absence of this line implies the default setting of annexb=yes (silence

suppression enabled). Thus, in order to interwork with this codec setting, the Avaya Communication Manager was set to G.729AB which is equivalent to G.729A with silence suppression.

- Call Park is not supported.

7. Verification Steps

The following steps may be used to verify the configuration:

- From the Avaya Communication Manager SAT, use the **status signaling-group** command to verify that the SIP signaling group is in-service.
- From the Avaya Communication Manager SAT, use the **status trunk-group** command to verify that the SIP trunk group is in-service.
- From the Avaya SES web administration interface, verify that all endpoints connected to the Adit 3104, both analog and SIP, are registered with the Avaya SES.
- Verify that calls can be placed to/from the analog and SIP endpoints connected to the Adit 3104.
- For further troubleshooting, logging of the SIP traffic can be enabled on the Adit 3104 by navigating to **System Monitoring → SIP Log** and checking the check box labelled ***Enabled***.

8. Support

For technical support on the Adit 3104 IP Business Gateway, contact Carrier Access toll-free at (800) 786-9929. Support can also be obtained via email at tech-support@carrieraccess.com or via the web site www.carrieraccess.com.

9. Conclusion

These Application Notes describe the procedures required to configure the Carrier Access Adit 3104 IP Business Gateway to interoperate with Avaya SIP Enablement Services and Avaya Communication Manager.

10. Additional References

[1] *Feature Description and Implementation For Avaya Communication Manager*, Doc # 555-245-205, Issue 4.0, February 2006.

[2] *Administrator Guide for Avaya Communication Manager*, Doc # 03-300509, Issue 2.1, May 2006

[3] *Installing and Administering SIP Enablement Services R3.1*, Doc# 03-600768, Issue 1.5, February 2006

[4] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, version 6.0, Doc # 210-100-500, Issue 9, June 2005

[5] *Avaya IA 770 INTUITY AUDIX Messaging Application*, Doc # 11-300532, May 2005

[6] *Carrier Access Adit 3104 IP Business Gateway Installation Guide*

[7] *Carrier Access Adit 3104 IP Business Gateway Administration Guide*

Product documentation for Avaya products may be found at http://support.avaya.com.
Product documentation for the Carrier Access Adit 3104 IP Business Gateway may be found at http://www.carrieraccess.com.

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.