Avaya Solution & Interoperability Test Lab

Application Notes for Thomson ST2022 SIP Telephones with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Aura™ Communication Manager, Avaya Aura™ SIP Enablement Services, and Thomson ST2022 SIP Telephones. During compliance testing, Thomson ST2022 SIP Telephones successfully registered with SIP Enablement Services, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features such as conference, transfer, and hold.

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

1. Introduction

These Application Notes describe a solution comprised of Avaya Aura™ Communication Manager, Avaya Aura™ SIP Enablement Services, and Thomson ST2022 SIP Telephones. Communication Manager and SIP Enablement Services has the capability to extend advanced telephony features to SIP stations. These features can be extended to non-Avaya SIP telephones such as the Thomson ST2022 SIP Telephones.

1.1. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily on verifying call establishment on the ST2022 SIP Telephones and operations such as dialing methods (manual, re-dial, and phone book), hold, mute, transfer and conference. In addition, ST2022 SIP Telephones' interactions with SIP Enablement Services, Communication Manager, and Avaya SIP, H.323, and Analog telephones were also verified.

1.2. Support

For technical support on ST2022 SIP Telephones, contact Thomson's technical support at the following:

- Telephone - Obtain the country specific hotline from here:
  http://www.thomsonbroadbandpartner.com//telephony-solutions/support/contact-us.php

- E-mail - Submit a request for assistance from here:
  http://www.thomsonbroadbandpartner.com/telephony-solutions/thomson-telecom/contact-us.php

2. Reference Configuration

Figure 1 illustrates a sample configuration consisting of Communication Manager running on an Avaya S8300C Server with the Avaya G350 Media Gateway, SIP Enablement Services running on the S8500C Server, and the Thomson ST2022 SIP Telephones. For completeness, an Avaya 9630 SIP IP Telephone, an Avaya 9630 H.323 IP Telephone and an Avaya 6221 Analog Telephone were included to demonstrate calls between the ST2022 SIP Telephones and Avaya SIP, H.323 and Analog telephones. The Avaya C364T-PWR Converged Stackable Switch provides Ethernet connectivity and power to the Avaya and Thomson SIP and H.323 Telephones through Power-over-Ethernet (PoE). Avaya Aura™ Communication Manager Messaging running on the S8300C Server is used to support voice messaging. An audio wav file is used as the music-on hold (MOH) through the virtual Voice Announcement with LAN (VAL) feature in the Avaya G350 Media Gateway. The ISDN-BRI trunk is also included to demonstrate calls routed by Communication Manager between the ST2022 SIP Telephones and the PSTN.

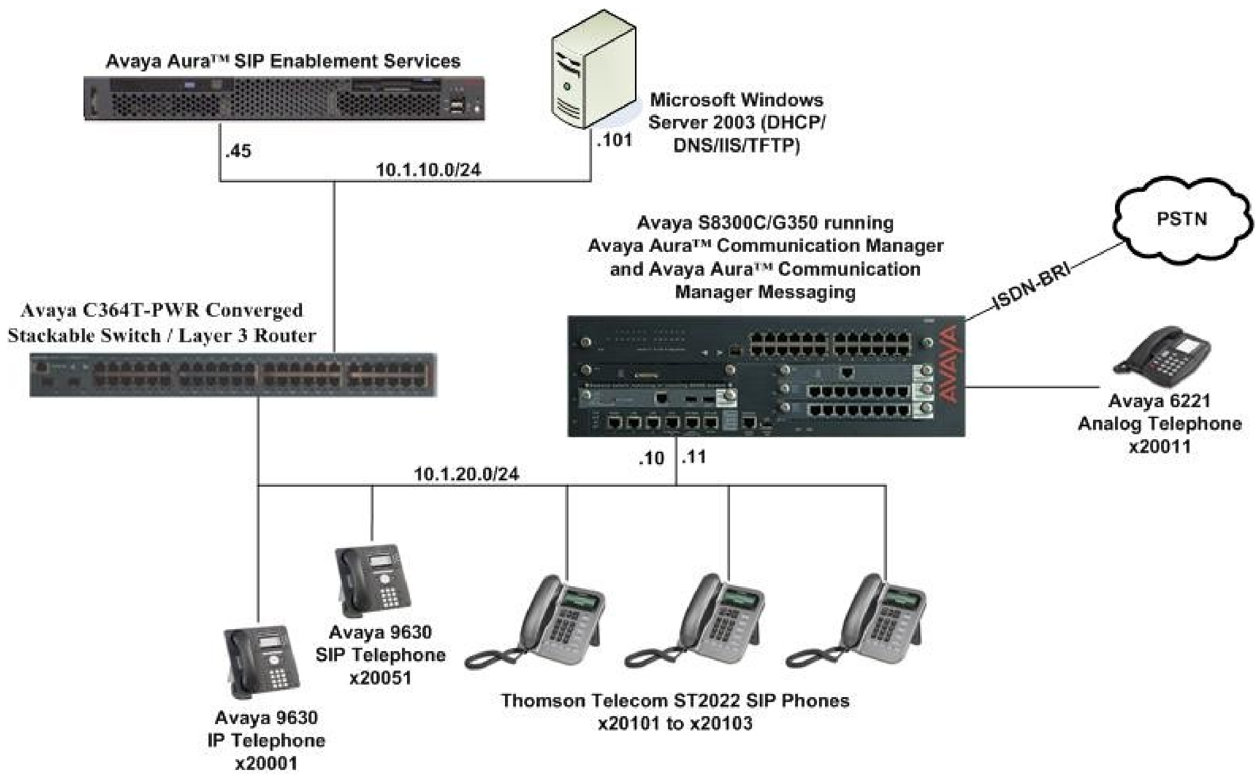


Figure 1: Sample Configuration

The ST2022 SIP Telephone originates a call by sending a call request (SIP INVITE message) to the SIP Enablement Services, which then routes the call over a SIP trunk to Communication Manager for origination services. If the call is destined for another local SIP telephone, then Communication Manager routes the call back over the SIP trunk to SIP Enablement Services for delivery to the destination SIP telephone. Otherwise, Communication Manager routes the call to the PSTN, a local Avaya H.323, digital, or analog telephone, as appropriate depending on the destination number.

For a call arriving at Communication Manager that is destined for the Thomson SIP Telephone, Communication Manager routes the call over the SIP trunk to the SIP Enablement Services for delivery to the Thomson SIP Telephone.

These application notes assume that Communication Manager and SIP Enablement Services are already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult [1] thru [5].

3. Equipment and Software Validated

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

Equipment	Software / Firmware
Avaya S8300C Server	Avaya Aura™ Communication Manager and Avaya Aura™ Communication Manager Messaging 5.2 (Service Pack 02.0.947.3-17579)
Avaya G350 Media Gateway	29.24.4
Avaya S8500C Server	Avaya Aura™ SIP Enablement Services 5.2 (Service Pack SES-02.0.947.3-SP2a)
Avaya C364T-PWR Converged Stackable Switch	4.5.18
Avaya 9630 IP Telephones	3.002 (H.323) 2.4.1 (SIP)
Avaya 6221 Analog Telephone	-
Thomson ST2022 SIP Telephones	H/W Version: V2 Boot Version: V3.03 DSP Version: V3.20 APP Version: V4.68

4. Configure Communication Manager

This section describes a procedure for setting up a SIP trunk between Communication Manager and SIP Enablement Services which includes steps for setting up a list of IP codecs, an IP network region, a signaling group and its interface. Before a trunk can be configured, it is necessary to verify if there is enough capacity to setup an additional trunk. Also, a procedure is described here to configure SIP telephones in Communication Manager. Configuration in the following sections is only for the fields where a value needs to be entered or modified. Default values are used for all other fields.

These steps are performed from the Communication Manager System Access Terminal (SAT) interface. ST2022 and other SIP telephones are configured as Outboard-Proxy SIP (OPS) Stations in Communication Manager. Communication Manager does not directly control an OPS endpoint, but its features and calling privileges can be applied to it by associating a local extension with the OPS endpoint. Similarly, a SIP telephone in SIP Enablement Services is associated with an extension on Communication Manager. SIP telephones register with the SIP Enablement Services and use Communication Manager for call origination and termination services. Enter the **save translation** command after completing this section.

4.1. Capacity Verification

Step	Description
1.	Enter the **display system-parameters customer-options** command. Verify that there are sufficient **Maximum Off-PBX Telephones – OPS** licenses. If not, contact an authorized Avaya account representative to obtain additional licenses.

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

     G3 Version: V15                             Software Package: Standard
       Location: 2                            RFA System ID (SID): 1
       Platform: 13                           RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 900   244
                                      Maximum Stations: 450   156
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 450   0
                    Maximum Off-PBX Telephones -   OPS: 450   5
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
```

2.	Proceed to **Page 2** of **OPTIONAL FEATURES** form. Verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses. **Note:** Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted.

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

IP PORT CAPACITIES                                            USED
                    Maximum Administered H.323 Trunks: 200    66
          Maximum Concurrently Registered IP Stations: 450    1
            Maximum Administered Remote Office Trunks: 450    0
Maximum Concurrently Registered Remote Office Stations: 450   0
             Maximum Concurrently Registered IP eCons: 2      0
  Max Concur Registered Unauthenticated H.323 Stations: 200   0
                 Maximum Video Capable H.323 Stations: 200    0
                  Maximum Video Capable IP Softphones: 200    0
                      Maximum Administered SIP Trunks: 450    20
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                            Maximum TN2501 VAL Boards: 0      0
                    Maximum Media Gateway VAL Sources: 2      1
          Maximum TN2602 Boards with 80 VoIP Channels: 0      0
         Maximum TN2602 Boards with 320 VoIP Channels: 0      0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

4.2. IP Codec Set

This section describes the steps for administering an IP codec set in Communication Manager. This codec set is used in the IP network region for communications between Communication Manager and SIP Enablement Services.

Step	Description
1.	Enter the **change ip-codec-set n** command, where **n** is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 4.3** for configuring an IP network region to specify which codec sets may be used within and between network regions. For the compliance testing, **G.711MU**, **G.711A** and **G.729A** were used and **Media Encryption** was set to **none**. Note also the value for **Packet Size (ms)** for each codec which should match the values configured on the ST2022 SIP telephones in **Section 6 Step 6**.

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

                          IP Codec Set

    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2         20
 2: G.711A            n         2         20
 3: G.729A            n         2         20
 4:
 5:
 6:
 7:


     Media Encryption
 1: none
 2:
 3:
```

4.3. IP Network Region

This section describes the steps for administering an IP network region in Communication Manager for communication between Communication Manager and SIP Enablement Services.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Enter the change ip-network-region n command, where n is a number between 1 and 250 inclusive and configure the following:
• Authoritative Domain – Set to sglab.com in this example. This should match the SIP Domain value configured in SIP Enablement Services Section 5 Step 2.
• Intra-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or SIP Enablement Services in the same IP network region.
• Inter-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or SIP Enablement Services in different IP network regions.
• Codec Set – Set the codec set number as provisioned in Section 4.2.
• Audio PHB Value – Note down the value to configure the ST2022 SIP telephone in Section 6 Step 5.
• Audio 802.1p Priority – Note down the value to configure the ST2022 SIP telephone in Section 6 Step 5.
<pre>
change ip-network-region 2   Page   1 of 19
                               IP NETWORK REGION
  Region: 2
Location: Authoritative Domain: sglab.com
    Name: Local
MEDIA PARAMETERS   Intra-region IP-IP Direct Audio: yes
      Codec Set: 2   Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048   IP Audio Hairpinning? n
   UDP Port Max: 65535
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: 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
</pre></td></tr>
</table>

2.	Proceed to **Page 3** of IP network region configuration and enable inter-region connectivity between regions as per below. For this compliance testing, codec set was set to the IP codec set configured in **Section 4.2**.

```
change ip-network-region 2                                      Page   3 of  19

                   Inter Network Region Connection Management

 src dst codec direct   WAN-BW-limits   Video      Intervening      Dyn
 rgn rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC IGAR AGL
 2   1    2      y   NoLimit                                             n
 2   2    2                                                                  all
 2   3
 2   4
 2   5
 2   6
 2   7
 2   8
 2   9
 2   10
 2   11
 2   12
 2   13
 2   14
 2   15
```

4.4. IP Node Names

This section describes the steps for administering a node name in Communication Manager for SIP Enablement Services to be used in the configuration of the SIP signaling group.

Step	Description
1.	Use the **change node-names ip** command to add a new node name for SIP Enablement Services.

```
change node-names ip                                            Page   1 of   2
                                 IP NODE NAMES
    Name              IP Address
default             0.0.0.0
msgserver           10.1.20.12
procr               10.1.20.10
ses1                10.1.10.45
```

4.5. SIP Signaling

This section describes the steps for administering a signaling group in Communication Manager for communication between Communication Manager and SIP Enablement Services.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">1.</td><td>Enter the command add signaling-group n, where n is an available signaling group and configure the following:

Group Type – Set to sip.
Transport Method – Set to tls.
Near-end Node Name - Set to procr.
Near-end Listen Port - Defaults to 5061 for TLS.
Far-end Node Name - Set to the node name configured in Section 4.4.
Far-end Listen Port - Defaults to 5061 for TLS.
Far-end Network Region - Set to the Region configured in Section 4.3.
Far-end Domain - Set to sglab.com in this example. This should match the SIP Domain value configured in SIP Enablement Services in Section 5 Step 2.
</td></tr>
<tr><td><pre>
add signaling-group 51
                                SIGNALING GROUP

 Group Number: 51   Group Type: sip
                        Transport Method: tls
  IMS Enabled? n

   Near-end Node Name: procr   Far-end Node Name: ses1
 Near-end Listen Port: 5061   Far-end Listen Port: 5061
                                          Far-end Network Region: 2
Far-end Domain: sglab.com

                                           Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload   Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3   IP Audio Hairpinning? n
         Enable Layer 3 Test? n   Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n   Alternate Route Timer(sec): 6
</pre></td></tr>
</table>

4.6. SIP Trunking

This section describes the steps for administering a trunk group in Communication Manager for communication between Communication Manager and SIP Enablement Services.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Issue the command add trunk-group n, where n is an unallocated trunk group and configure the following:

Group Type – Set to the Group Type field value configured in Section 4.4.
Group Name – Enter any descriptive name.
TAC (Trunk Access Code) – Set to any available trunk access code.
Signaling Group – Set to the Group Number field value configured in Section 4.5. (i.e., 51)
Number of Members – Allowed values are between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

Note: Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunk members for the duration of the call. The license file installed on the system controls the maximum permitted.</td></tr>
<tr><td></td><td><pre>
add trunk-group 51   Page   1 of 21
                                TRUNK GROUP

Group Number: 51   Group Type: sip   CDR Reports: n
  Group Name: SIP Endpoints   COR: 1   TN: 1   TAC: 751
   Direction: two-way   Outgoing Display? n
 Dial Access? n   Night Service:
Queue Length: 0
Service Type: tie   Auth Code? n

                                                      Signaling Group: 51
                                                    Number of Members: 20
</pre></td></tr>
<tr><td>2.</td><td>Proceed to Page 4 and set Telephone Event Payload Type to 96 to match the default value used by the ST2022 SIP telephones. Leaving this value blank is also acceptable as the Communication Manager and the ST2022 SIP telephones will negotiate the payload type.</td></tr>
<tr><td></td><td><pre>
add trunk-group 51   Page   4 of 21
                              PROTOCOL VARIATIONS

                        Mark Users as Phone? n
              Prepend '+' to Calling Number? n
        Send Transferring Party Information? n
                   Network Call Redirection? n
                      Send Diversion Header? n
                    Support Request History? y
             Telephone Event Payload Type: 96
</pre></td></tr>
</table>

4.7. SIP Stations

This section describes the steps for administering OPS stations in Communication Manager and associating the OPS station extensions with the telephone numbers of the ST2022 SIP telephones.

Step	Description
1.	Enter the **add station n** command, where **n** is an available extension in the dial plan, to administer an OPS station. On Page 1 of the **STATION** form configure the following fields: • **Type** – Set to **4620**. • **Port** – Set to **X**. • **Name** – Enter any descriptive name.
	(screen below)

```
add station 20101                                              Page   1 of   5
                                     STATION

Extension: 20101                         Lock Messages? n              BCC: 0
     Type: 4620                          Security Code:                 TN: 1
     Port: X                           Coverage Path 1:                COR: 1
     Name: John Doe                    Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 20101
            Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english               Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal             Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n

                                              Customizable Labels? y
```

2.	Proceed to **Page 2** of the **STATION** form and set **Restrict Last Appearance** to **n**.

```
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe            Auto Select Any Idle Appearance? n
          LWC Activation? y                       Coverage Msg Retrieval? y
  LWC Log External Calls? n                                  Auto Answer: none
             CDR Privacy? n                             Data Restriction? n
   Redirect Notification? y                   Idle Appearance Preference? n
 Per Button Ring Control? n                 Bridged Idle Line Preference? n
   Bridged Call Alerting? n                     Restrict Last Appearance? n
  Active Station Ringing: single
                                                         EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed                      EC500 State: disabled
         Multimedia Mode: enhanced
    MWI Served User Type: qsig-mwi            Display Client Redirection? n
                                             Select Last Used Appearance? n
                                               Coverage After Forwarding? s

                                             Direct IP-IP Audio Connections? y
  Emergency Location Ext: 20101        Always Use? n IP Audio Hairpinning? n
```

3.	Proceed to **Page 4** of the **STATION** form and add the required number of **call-appr** entries in the **BUTTON ASSIGNMENTS** section. The number of call appearances should match the **Call Limit** field value in **Step 5**. Configure additional feature buttons such as **no-hld-cnf** (required for Conference on Answer) and **auto-cback** (required for Automatic Call Back) as required.

```
add station 20101                                             Page   4 of   5
                                     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: no-hld-cnf
 2: call-appr                               6: auto-cback
 3:                                         7:
 4:                                         8:
```

4.	Enter the **change off-pbx-telephone configuration-set n** command, where **n** is an unused configuration set to be used for the ST2022 SIP telephones. On the **CONFIGURATION SET** form, configure the following fields: • **Configuration Set Description** – Set to a descriptive name. • **Calling Number Style** – Set to the recommended value of **network**. Use the default values for the remaining fields. For the detail explanation of each field, refer to [5].

```
change off-pbx-telephone configuration-set 1                    Page   1 of   1



                                  CONFIGURATION SET: 1

                      Configuration Set Description: SIP Phones
                               Calling Number Style: network
                                CDR for Origination: phone-number
                   CDR for Calls to EC500 Destination? y
                          Fast Connect on Origination? n
                          Post Connect Dialing Options: dtmf
                        Cellular Voice Mail Detection: none
                                        Barge-in Tone? n
                          Calling Number Verification? y
            Call Appearance Selection for Origination: primary-first
                                    Confirmed Answer? n
```

5.	Enter the **add off-pbx-telephone station-mapping** command and configure the following: • **Station Extension** – Set the extension of the OPS station as configured above. • **Application** – Set to **OPS**. • **Phone Number** – Enter the number that the ST2022 SIP telephone will use for registration and call termination. In the example below, the **Phone Number** is the same as the **Station Extension**, though it is not required to be the same. • **Trunk Selection** – Set to the trunk group number configured in **Section 4.6**. • **Config Set** – Set to the configuration set configured in **Step 4**.

```
add off-pbx-telephone station-mapping                           Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial    CC  Phone Number     Trunk        Config  Dual
 Extension                 Prefix                        Selection    Set     Mode
 20101             OPS          -      20101              51           1
```

6.	Proceed to **Page 2** of station mapping form and verify that the **Call Limit** field value matches the number of call appearances configured in **Step 2**.	

```
add off-pbx-telephone station-mapping                          Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 20101         OPS      2        both        all         both
```

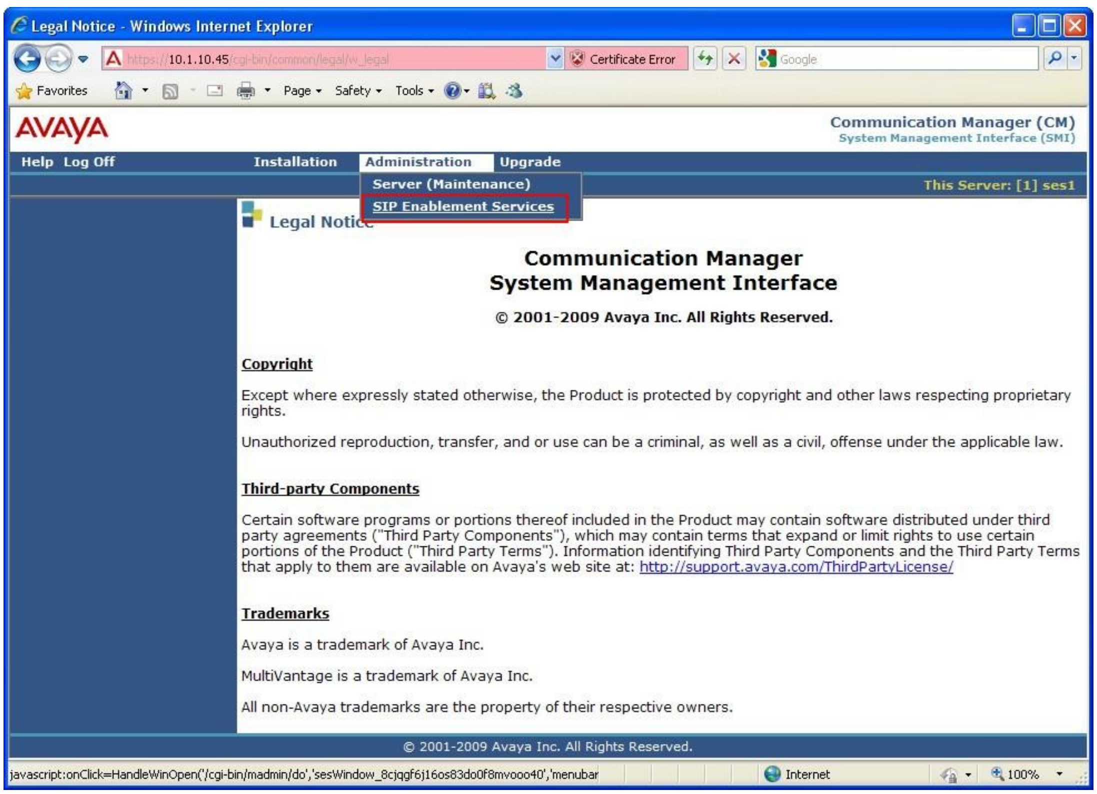
7.	Repeat **Steps 1 - 6** as necessary to administer additional OPS stations and associations for the ST2022 SIP telephones.

5. Configure SIP Enablement Services

This section describes the steps for creating a SIP trunk between SIP Enablement Services and Communication Manager. Also, SIP user accounts are configured in SIP Enablement Services and associated with a Communication Manager OPS station extension. The ST2022 SIP telephones will register with SIP Enablement Services using the SIP user accounts.

The configuration in the following steps is only for the fields where a value needs to be entered or modified. Default values are used for all other fields.

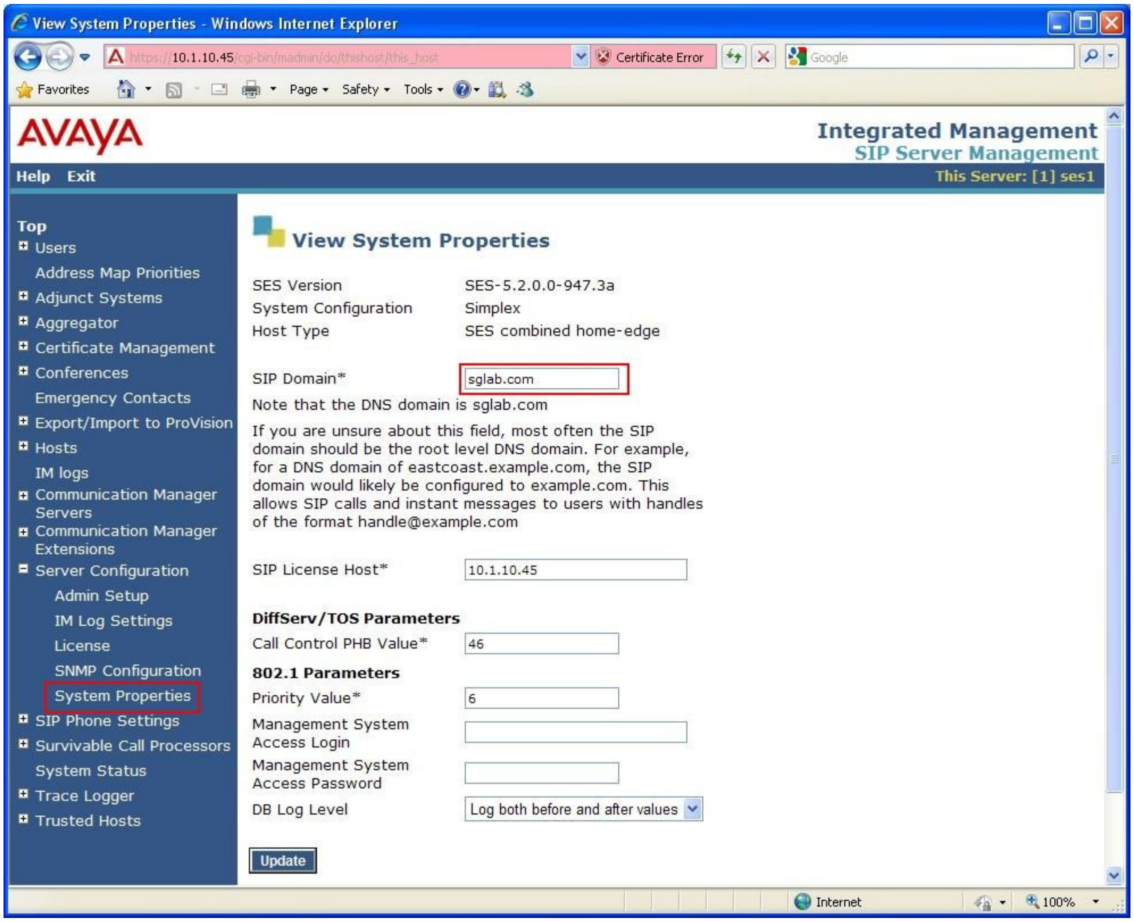
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Open a web browser, enter http://<IP address of SIP Enablement Services Server>/admin for the URL, and log in with the appropriate credentials (not shown). Click on Administration > SIP Enablement Services upon successful login.



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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.</td><td>On the SIP Server Management page:

Click the + sign to expand the options under Server Configuration.
Click System Properties.
Verify the SIP Domain matches the Far-end Domain field value configured for the signaling group on Communication Manager in Section 4.5.



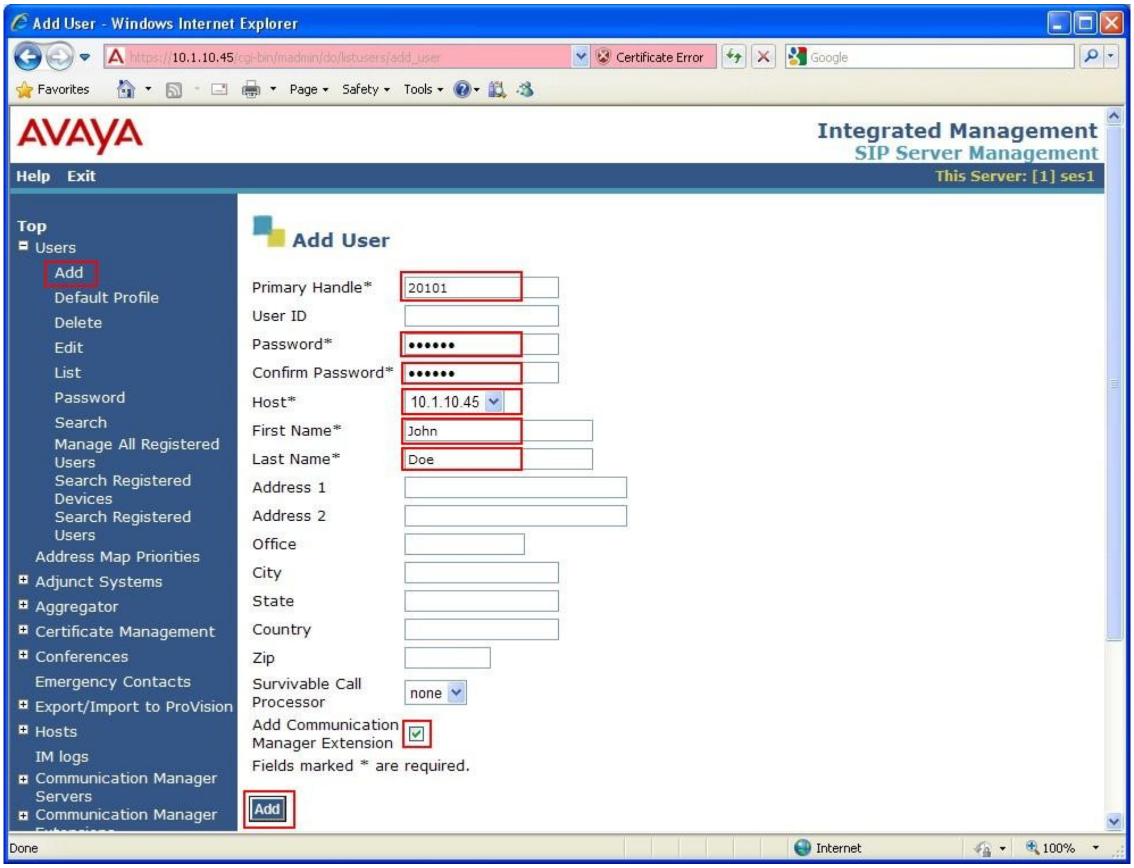
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>

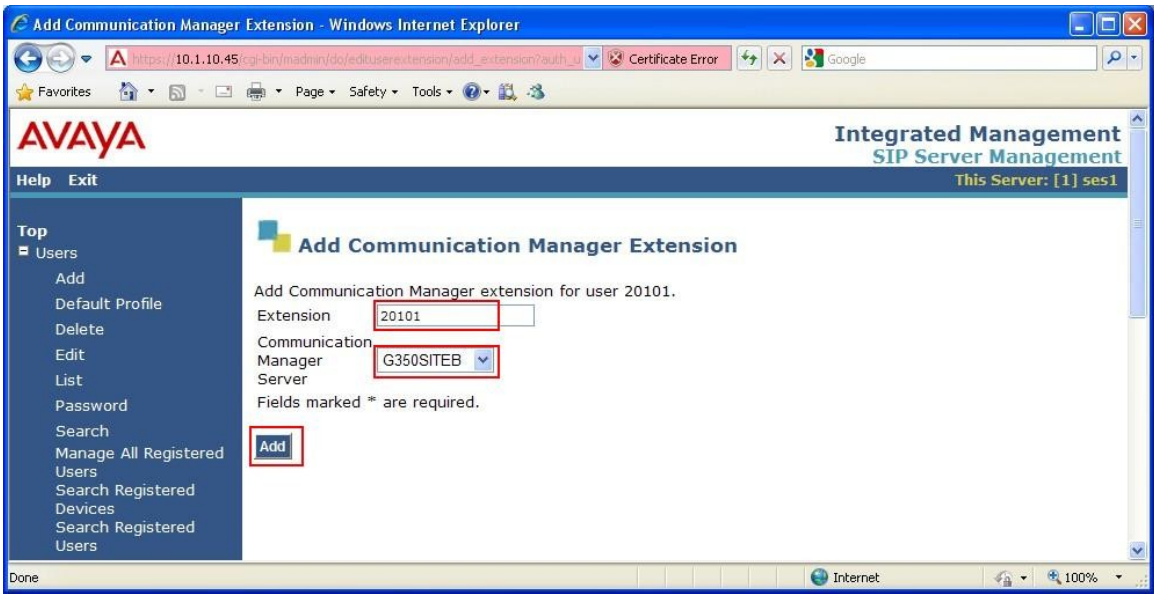
In the left pane of the **SIP Server Management** page, expand **Users** and click **Add**. At the **Add User** page, configure the following:

- **Primary Handle –** Enter the phone number of the ST2022 SIP telephone. This number was configured in **Section 4.7 Step 1**.
- **User ID** – Set to any descriptive name (optional).
- **Password** and **Confirm Password –** Specify a password that the ST2022 SIP telephone will use to register with SIP Enablement Services.
- **Host –** Select the IP address of the SIP Enablement Services server.
- **First Name** and **Last Name –** Enter descriptive names.
- Check the **Add Communication Manager Extension** checkbox.

Click **Add** when finished and then click **Continue** on the next page [not shown].



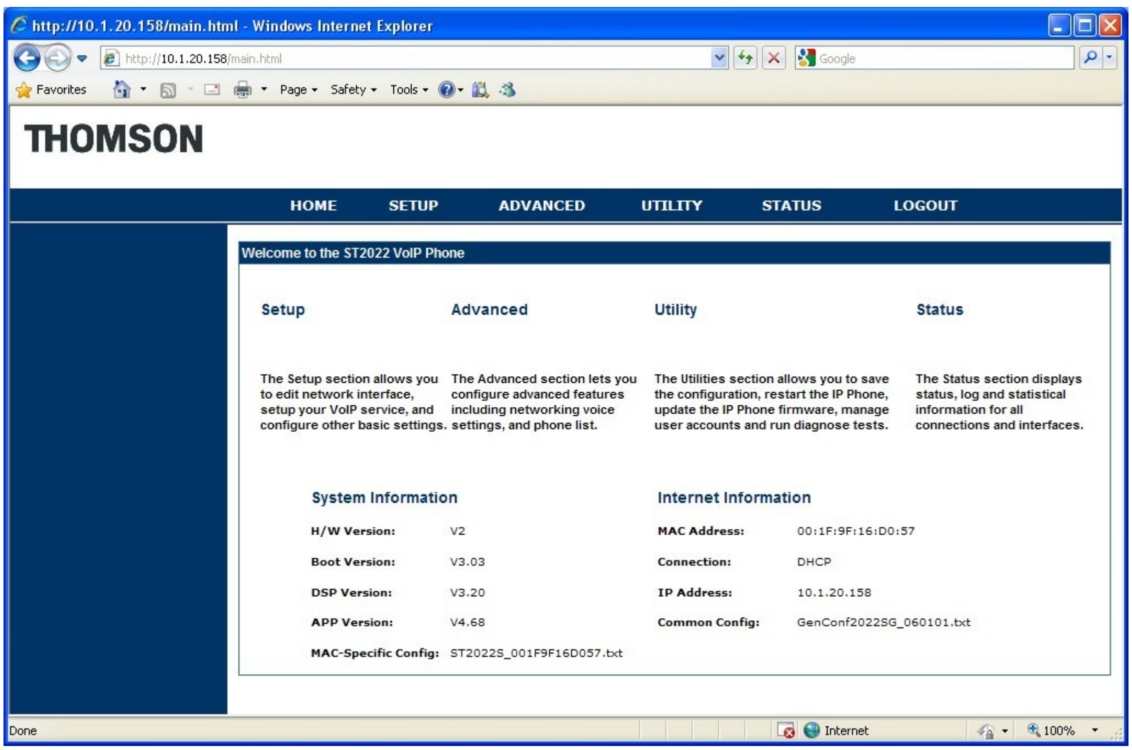
</td></tr>
</table>

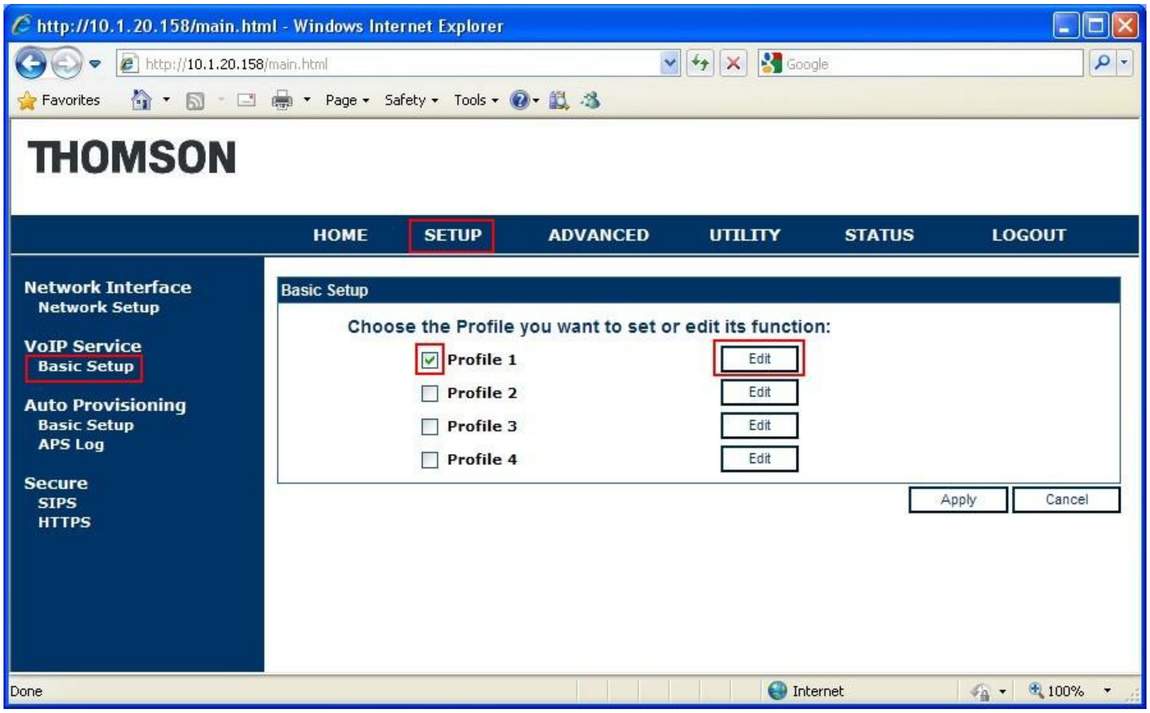
Step	Description
4.	At the **Add Communication Manager Extension** page, configure the following: • **Extension –** Set to **Phone Number** field value configured in **Section 4.7 Step 1**. • **Communication Manager Server** – Set to the Communication Manager where this OPS station is configured. • Click **Add** and then click **Continue** on the next page [not shown]. **Note:** Communication Manager Server was previously configured during the initial setup of SIP Enablement Services. 
5.	Repeat **Steps 3** and **4** as necessary to configure additional ST2022 SIP telephones.

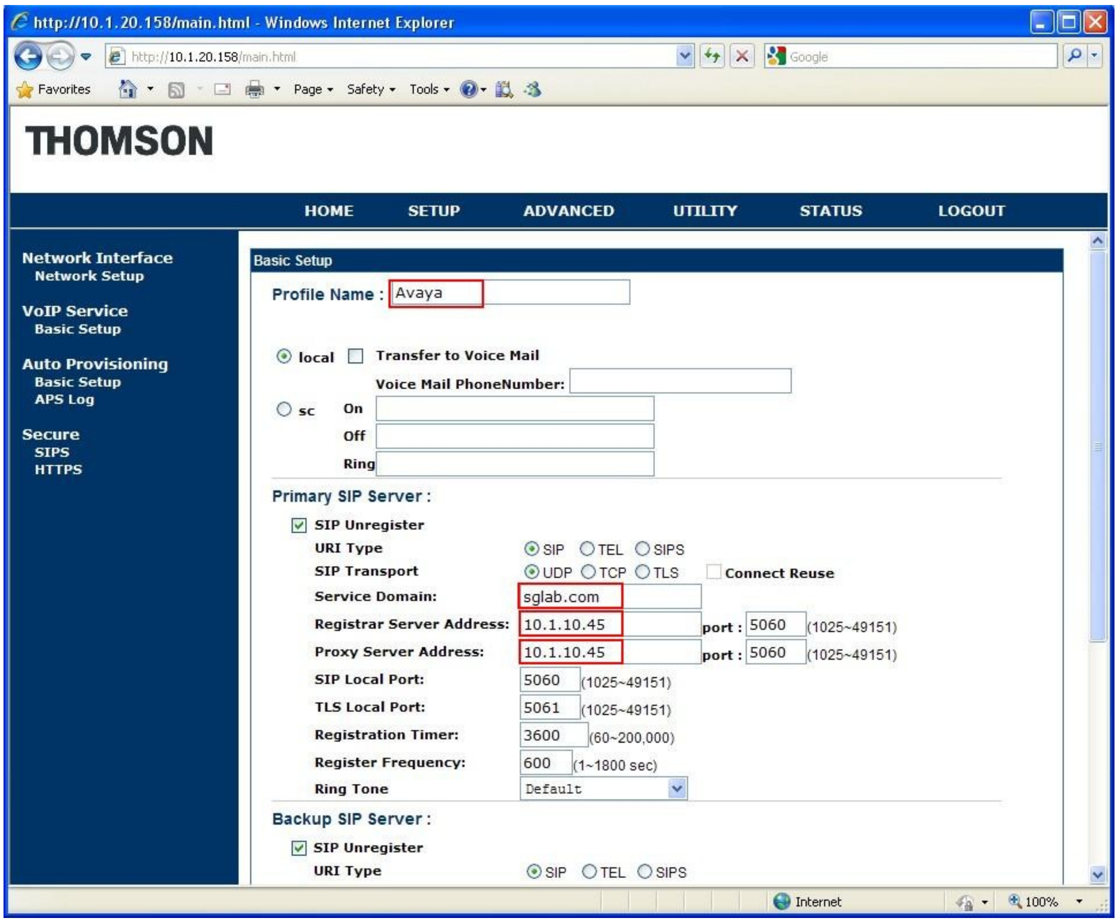
6. Configure Thomson ST2022 SIP Telephones

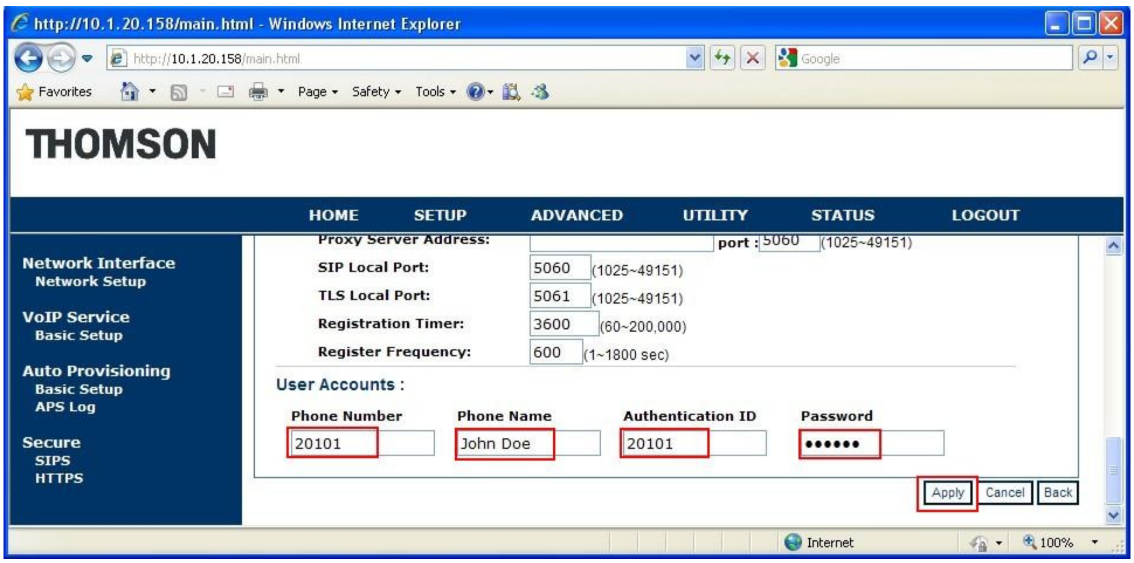
This section describes the steps for configuring the Thomson ST2022 SIP Telephones. The ST2022 SIP telephones support DHCP and automatic provisioning. For the compliance testing, the ST2022 SIP telephones were assigned IP addresses using DHCP but were manually configured via the ST2022 SIP telephones' web interface. Some of the configuration requires a restart of the telephone to take effect and thus should be followed as instructed.

Step	Description
1.	Open a web browser and enter **http://<a.b.c.d>/admin.html** for the URL, where **a.b.c.d** is the IP address of the ST2022 SIP telephone and log in using an account with administrative privileges and the home screen will be displayed as shown below.



Step	Description
2.	Click **SETUP** from the top menu and then click **Basic Setup** from the left menu. The Basic Setup page will be shown. The ST2022 SIP telephone supports profiles to ease the configuration of SIP registration. Check an unused profile and click **Edit** to configure the profile. 

Step	Description
3.	Configure the profile as follows for the ST2022 SIP telephone to register to SIP Enablement Services: • **Profile Name** – Enter a descriptive name. • **Service Domain –** Set to the **SIP Domain** of the SIP Enablement Services as shown in **Section 5 Step 2**. • **Registrar Server Address** – Set to the IP address of the SIP Enablement Services. • **Proxy Server Address –** Set to the IP address of the SIP Enablement Services. The remaining fields may be left at their default values. 

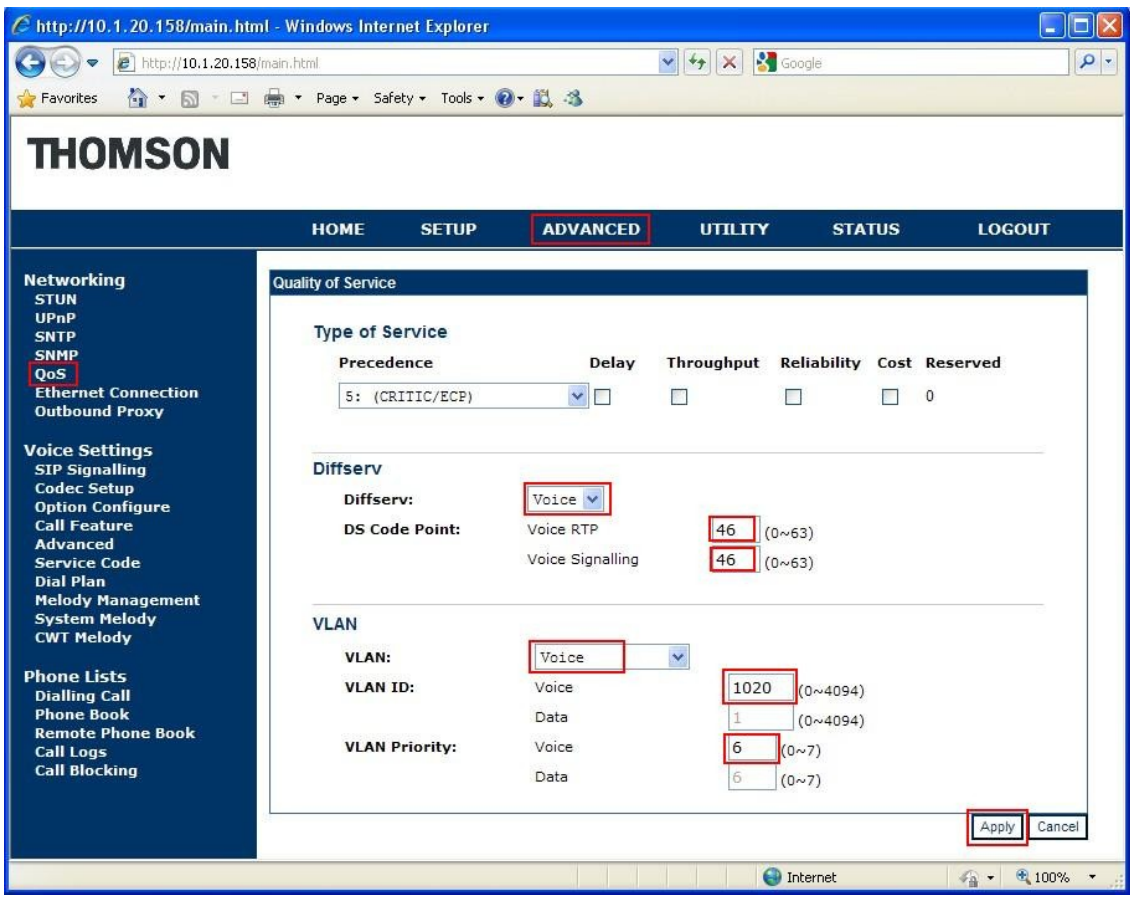
Step	Description
4.	Scroll to the bottom of the page and configure the User Accounts section as follows: • **Phone Number** – Set to the **Primary Handle** of a SIP User on SIP Enablement Services as configured in **Section 5 Step 3**. • **Phone Name –** Enter a name for the user. • **Authentication ID –** Set to the **Primary Handle** of a SIP User on SIP Enablement Services as configured in **Section 5 Step 3**. • **Password –** Set to the **Password** of the SIP User as configured in **Section 5 Step 3**. Click **Apply** to continue. 

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

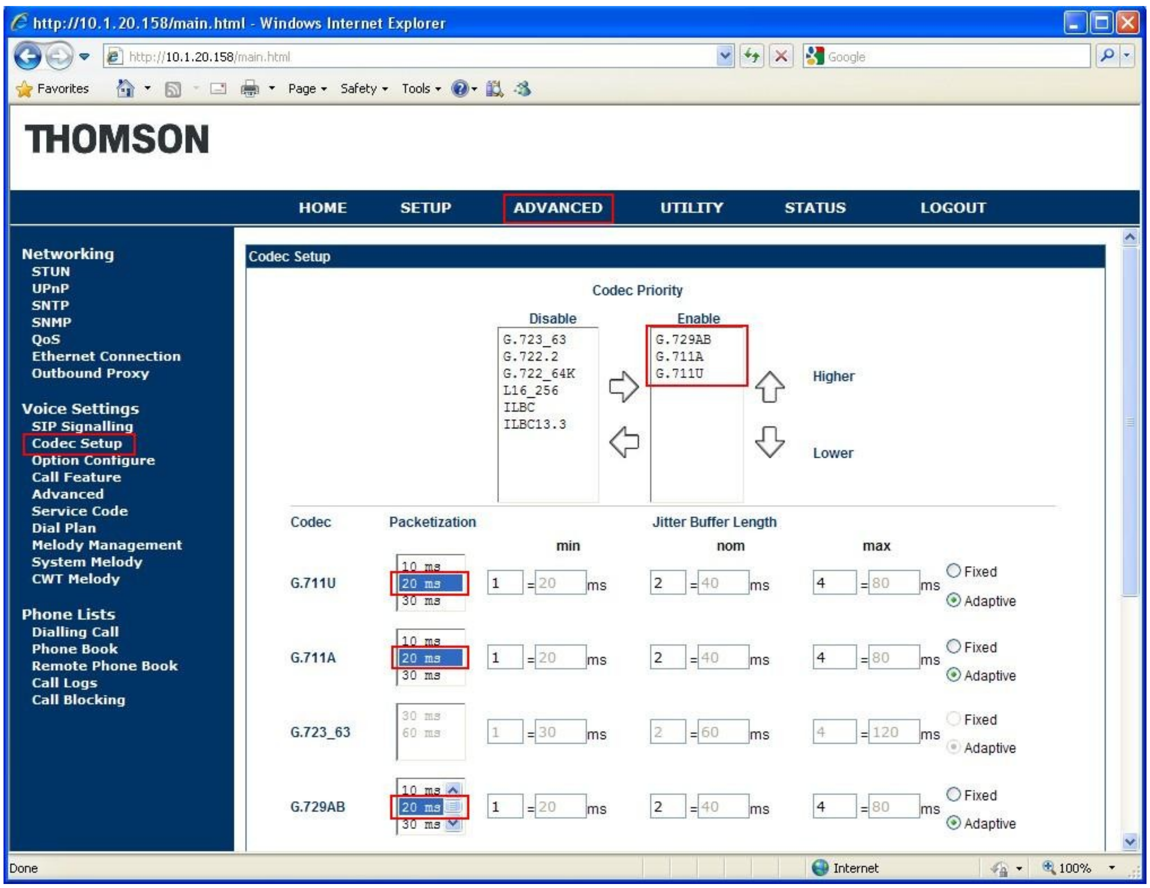
Click **ADVANCED** from the top menu and then click **QoS** from the left menu. The Quality of Service page will be shown. Configure the **Diffserv** and **VLAN** section as follows:

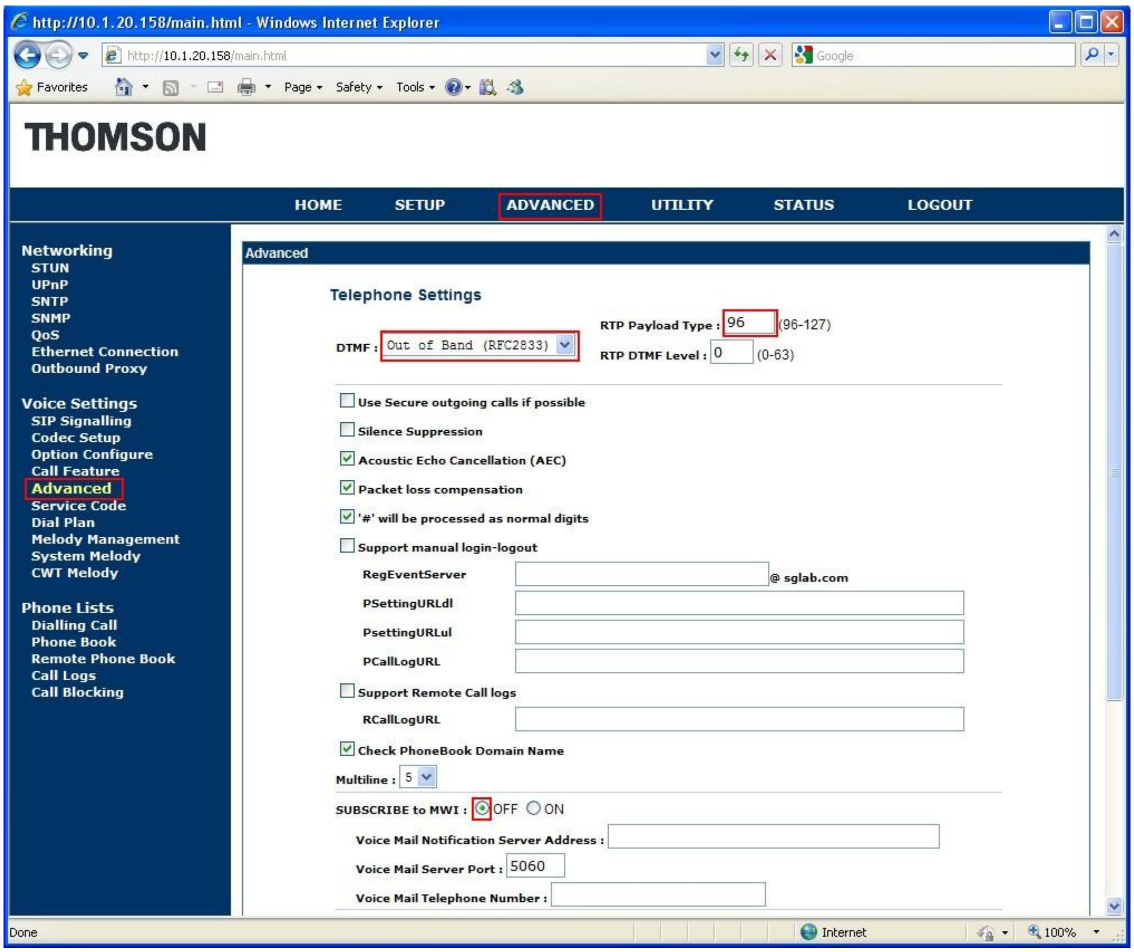
- **Diffserv** – Select **Voice** to enable the Diffserv feature.
- **Voice RTP –** Set to the **Audio PHB Value** field value configured on Communication Manager in **Section 4.3 Step 1**.
- **Voice Signaling –** Set to the **Call Control PHB Value** field value configured on Communication Manager in **Section 4.3 Step 1**.
- **VLAN –** Select **Voice** to enable VLAN tagging.
- **VLAN ID > Voice -** Set to an appropriate VLAN tag used on the Ethernet switch.
- **VLAN Priority > Voice -** Set to the **Audio 802.1p Priority** field value configured on Communication Manager in **Section 4.3 Step 1**. This allows the Ethernet switch to prioritize voice packets over other data packets on the network.

Click **Apply** to continue.



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

Step	Description
6.	Click **ADVANCED** from the top menu and then click **Codec Setup** from the left menu. The Codec Setup page will be shown. Configure the **Codec Priority** and **Codec** section as follows: • **Codec Priority > Enable** – Use the arrow buttons to enable the **G.711U**, **G.711A** and **G.729AB** codecs that are used in this test configuration. • **Codec > Packetization** - Set to 20 ms for the codecs **G.711U**, **G.711A** and **G.729AB** to match the values configured on Communication Manager in **Section 4.2 Step 1**. Scroll to the bottom of the page and click **Apply**. 

Step	Description
7.	Click **ADVANCED** from the top menu and then click **Advanced** from the left menu. The Advanced page will be shown. Configure the following as shown below: • **DTMF** – Select **Out of Band (RFC2833)**. • **RTP Payload Type** - Accept the default value of **96** or set a value to match the **Telephone Event Payload Type** field value configured on Communication Manager in **Section 4.6 Step 2**. • **SUBSCRIBE to MWI** - Select **OFF**. Scroll to the bottom of the page and click **Apply**. This completes the configuration of the Thomson ST2022 SIP Telephone for basic operation. 

7. General Test Approach and Test Results

The general test approach was to place calls to and from the ST2022 SIP telephones and exercise basic telephone operations. The main objectives were to verify that:

- ST2022 SIP telephones successfully register with SIP Enablement Services.
- ST2022 SIP telephones successfully establish calls with Avaya SIP, H.323, and Analog telephones attached to SIP Enablement Services or Communication Manager.
- ST2022 SIP telephones successfully establish calls with PSTN telephones through Communication Manager.
- ST2022 SIP telephones successfully handle concurrent calls.
- ST2022 SIP telephones successfully negotiate the right codec.
- ST2022 SIP telephones successfully shuffle for VoIP calls.
- ST2022 SIP telephones successfully transmit DTMF during a call.
- ST2022 SIP telephones successfully hold and transfer a call.
- ST2022 SIP telephones establish a three-party conference call, and display calling party number.

All test cases were successfully completed.

8. Verification Steps

The following steps may be used to verify the configuration:

- From SIP Enablement Services Web Interface, verify that the ST2022 SIP telephones successfully register with SIP Enablement Services by using the **Users → Search Registered Users** link.
- Place calls to and from the ST2022 SIP telephones and verify that the calls are successfully established with two-way talk path.
- From the Communication Manager System Access Terminal (SAT) interface, perform the following steps to verify:
  - Audio codec used between two telephones
  - Shuffling between two telephones

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Enter status trunk n command, where n is the SIP trunk configured in Section 4.6. Note down the Member with Service State set to in-service/active. In this example, 0051/001 and 0051/003 are active and either member can be used to verify whether calls shuffled and which codec was used.
<pre>
status trunk 51   Page   1

                          TRUNK GROUP STATUS

Member   Port   Service State   Mtce Connected Ports
                                      Busy

0051/001 T00011   in-service/active   no   T00032
0051/002 T00012   in-service/idle   no
0051/003 T00013   in-service/active   no   T00011
0051/004 T00014   in-service/idle   no
</pre></td></tr>
</table>

2.	Enter **status trunk n,** where **n** is the member in active state as noted in the previous step for verification of codec used and shuffling status: • **Codec Type** – The codec used for Audio is **G.729** in this example. • **Shuffling** - If the **Near-end** and **Far-end** IP addresses for **Audio** belong to the ST2022 SIP telephones and the **Audio Connection Type** is **ip-direct**, it signifies that shuffling was successful. In this example, shuffling was successful.

```
status trunk 51/1                                                  Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: 01A0017
  Signaling    IP Address                              Port
   Near-end:  10.1.20.10                             : 5061
    Far-end:  10.1.10.45                             : 5061
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:          H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                        Codec Type: G.729
   Audio       IP Address                              Port
   Near-end:  10.1.20.159                            : 41000
    Far-end:  10.1.20.158                            : 41000

 Video Near:
  Video Far:
 Video Port:
  Video Near-end Codec:              Video Far-end Codec:
```

9. Conclusion

These Application Notes describe a solution comprised of Communication Manager, SIP Enablement Services and Thomson ST2022 SIP Telephones. During compliance testing, ST2022 SIP telephones successfully registered with SIP Enablement Services, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features like three-party conference, transfers, hold, etc. All test cases were successfully completed.

10. Additional References

Product documentation for Avaya products may be found at http://support.avaya.com/.

[1] *Administering Avaya Aura™ Communication Manager*, Release 5.2, Issue 5.0, May 2009, Document Number 03-300509.

[2] *Administering Network Connectivity on Avaya Aura™ Communication Manager*, Issue 14, May 2009, Document Number 555-233-504.

[3] *SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers*, Issue 9, May 2009, Document Number 555-245-206.

[4] *Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services*, Issue 7.0, May 2009, Document Number 03-600768.

[5] *Avaya Aura™ Communication Manager Screen Reference*, Release 5.2, Issue 1.0, May 2009, Document Number 03-602878.

Product information for Thomson products may be found at http://www.thomsonbroadbandpartner.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 DevConnect Program at devconnect@avaya.com.