Avaya Solution & Interoperability Test Lab

Application Notes for VTech CTM-S2210-X and CTM-S2211-SPK Corded SIP Hospitality Room Phones with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for VTech CTM-S2101/CTM-S2210-X/CTM-S2212 and CTM-S2211-X/CTM-S2211-SPK/CTM-S2213 Corded SIP Hotel Room Phones v3.3.0.8 and v3.4.0.0-9, respectively, to interoperate with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1. VTech CTM-S2101/CTM-S2210-X/CTM-S2212 and CTM-S2211-X/CTM-S2211-SPK/CTM-S2213 hospitality phones register with Avaya Aura® Session Manager as a SIP endpoint.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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 the configuration steps required for VTech CTM-S2101/CTM-S2210-X/CTM-S2212 and CTM-S2211-X/CTM-S2211-SPK/CTM-S2213 Corded SIP Hotel Room Phones to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. VTech SIP hospitality phones are available in 1- or 2-line styles with corded and cordless handsets. VTech CTM-S2101/CTM-S2210-X/CTM-S2212 and CTM-S2211-X/CTM-S2211-SPK/CTM-S2213 Corded SIP Hotel Room Phones register to Avaya Aura® Session Manager as a SIP endpoint. Compliance testing used the VTech CTM-S2210-X/CTM-S2211-SPK 1-line corded SIP Hotel Room Phones as representative models. See **Attachment 1** which provides details of VTech CTM-S2210-X Corded SIP Hotel Room Phone equivalency to the CTM-S2101 and CTM-S2212 SIP Hotel Phone models. See **Attachment 2** which provide details of VTech CTM-S2211-SPK Corded SIP Hotel Room Phone equivalency to the CTM-S2211-X and CTM-S2213 SIP Hotel Phone models.

2. General Test Approach and Test Results

The general test approach was to place calls to and from CTM-S2210-X/CTM-S2211-SPK and exercise basic telephone operations.

As the purpose of these phones is for hotel guest rooms, certain functionality considered to be standard on Avaya endpoints is not supported, and therefore, was not tested. For example, VTech CTM-S2210-X/CTM-S2211-SPK do not support transfers or conferences. More details on these limitations are described in the Test Results in **Section 2.2**.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and the VTech phones enabled capabilities of TLS/SRTP.

2.1. Interoperability Compliance Testing

The following areas were evaluated in the interoperability compliance test:

- Registration of CTM-S2210-X/CTM-S2211-SPK to Session Manager.
- Basic call features: Answer, Hold/Resume, Mute/Un-mute, Drop, Message Waiting Indicator, DTMF, Call Pickup, Call Waiting, and Call Forward.
- G.711 and G.729 codec support, codec negotiation, and Session Refresh Interval.
- Media (Shuffling) enabled and disabled.
- TLS transport and SRTP media encryption.
- Hospitality features: Automatic Wakeup Call and Housekeeping status.
- Serviceability testing to validate recovery from network connectivity loss.

2.2. Test Results

All test cases were executed. The following observations were made during the testing:

- CTM-S2210-X/CTM-S2211-SPK does not support the following features
    - Call Park/unpark
    - Transfer
    - Conference
    - Programmable buttons do not support feature access codes with data input
- The VTech G.726 codec payload type of 2 is not supported by Communication Manager. This is acceptable to VTech.
- CTM-S2210-X/CTM-S2211-SPK does not support SDP negotiation capabilities per (RFC5939) between SRTP and non-SRTP modes so codec sets for the phones must not offer both modes. Employing a separate codec set for the phones' encryption capabilities is a possible alternative to support endpoints that use SDP negotiation in a mixed environment.
- The web administration setting **Only accept trusted certificates** must be set for CTM-S2210-X/CTM-S2211-SPK to validate Session Manager's identity certificate during TLS session setup. Subsequent access to the web administration's **Trusted Certificates** page will not show that it has been set. This will be fixed in a future release of CTM-S2210-X/CTM-S2211-SPK firmware.

2.3. Support

Technical support for VTech CTM-S2101/CTM-S2210-X/CTM-S2212 and CTM-S2211-X/CTM-S2211-SPK/CTM-S2213 SIP Phones can be obtained at:

- Phone: 1 (888) 907-2007
- https://vtechhotelphones.com

3. Reference Configuration

Figure 1 illustrates the test configuration diagram for CTM-S2210-X/CTM-S2211-SPK integrated with Avaya Aura® Communication Manager and Avaya Aura® System Manager.

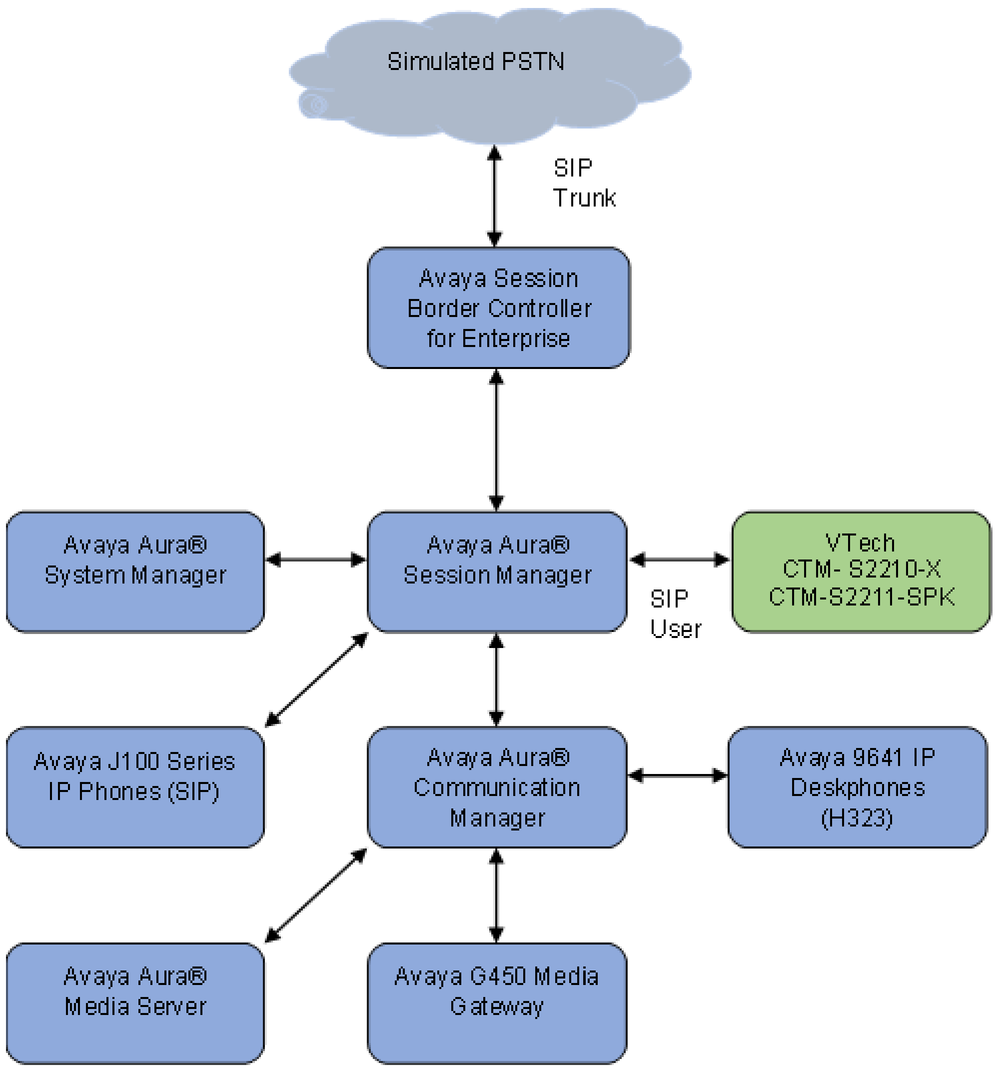


Figure 1: Avaya Test Configuration for CTM-S2210-X/CTM-S2211-SPK Corded SIP Hotel Room Phones

4. Equipment and Software Validated

The following equipment and software were used for the compliance test provided:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Virtual Machine	10.1.0.2-SP2 01.0.974.0-27607
Avaya Aura® System Manager running on Virtual Machine	10.1.0.2 Service Pack 2 10.1.0.2.0715160
Avaya Aura® Session Manager running on Virtual Machine	10.1.0.2 Service Pack 2 10.1.0.02.1010215
Avaya Session Border Controller for Enterprise running on Virtual Machine	10.1.1.0-35-21872
Avaya Aura® Media Server running on Virtual Machine	10.1.0.101
Avaya G450 Media Gateway	42.7.0
Avaya 9641G IP DeskPhone	6.8532 (H.323)
Avaya J179 IP Phone	4.0.13.0.6 (SIP)
VTech CTM-S2210-X Corded SIP Hotel Phone	3.3.0.8
VTech CTM-S2211-SPK Corded SIP Hotel Phone	3.4.0.0-9

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager. The procedures include the following areas:

- Verify Communication Manager OPS Licensed Capacity
- Administer IP Network Region
- Administer IP Codec Set

Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials. The configuration steps illustrate field values changed for this reference configuration. Default values were used for all other fields.

Note: It is assumed that basic configuration of the Communication Manager has already been completed, such as the SIP trunk to Session Manager. The SIP station configuration for CTM-S2210-X/CTM-S2211-SPK is configured through System Manager in **Section 6.2.**

5.1. Verify Communication Manager OPS Licensed Capacity

Using the SAT, verify that the Off-PBX Stations (OPS) and SIP Trunks features are enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```
display system-parameters customer-options                     Page   1 of  12
                                OPTIONAL FEATURES

     G3 Version: V20                            Software Package: Enterprise
       Location: 2                               System ID (SID): 1
       Platform: 28                              Module ID (MID): 1

                                                          USED
                        Platform Maximum Ports:  48000    150
                              Maximum Stations:    150    115
                      Maximum XMOBILE Stations:  36000      0
              Maximum Off-PBX Telephones - EC500:    150      0
              Maximum Off-PBX Telephones -   OPS:    150     62
              Maximum Off-PBX Telephones - PBFMC:    150      0
              Maximum Off-PBX Telephones - PVFMC:    150      0
              Maximum Off-PBX Telephones - SCCAN:      0      0
                   Maximum Survivable Processors:  313        0

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

5.2. Administer IP Network Region

This IP network region is for the signaling group associated with the SIP trunk between Session Manager and Communication Manager. This form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. Verify the following values:

- **Authoritative Domain**: The applicable domain (e.g., *avaya.com*)
- **Codec Set**: The codec set number from **Section 5.3**

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 G450 Media Gateway or Media Server.

```
change ip-network-region 1                                          Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name: Main
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? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.3. Administer IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to CTM-S2210-X/CTM-S2211-SPK. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set *1* is specified in **IP Network Region** *1* from **Section 5.2**. The form shows the list of codecs tested. Enter values for the following:

- **Audio Codec**: The audio codecs tested
- **Media Encryption**: Do not include *none*

Note: Avaya endpoints supporting both RTP/SRTP may be administered in a separate codec set.

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

                         IP MEDIA PARAMETERS
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU          n          2        20
 2: G.711A           n          2        20
 3: G.729            n          2        20
 4:
 5:
 6
 7:
Media Encryption                  Encrypted SRTCP: best-effort
 1:1-srtp-aescm128-hmac80
 2:2-srtp-aescm128-hmac32
 3:
 4:
 5:
```

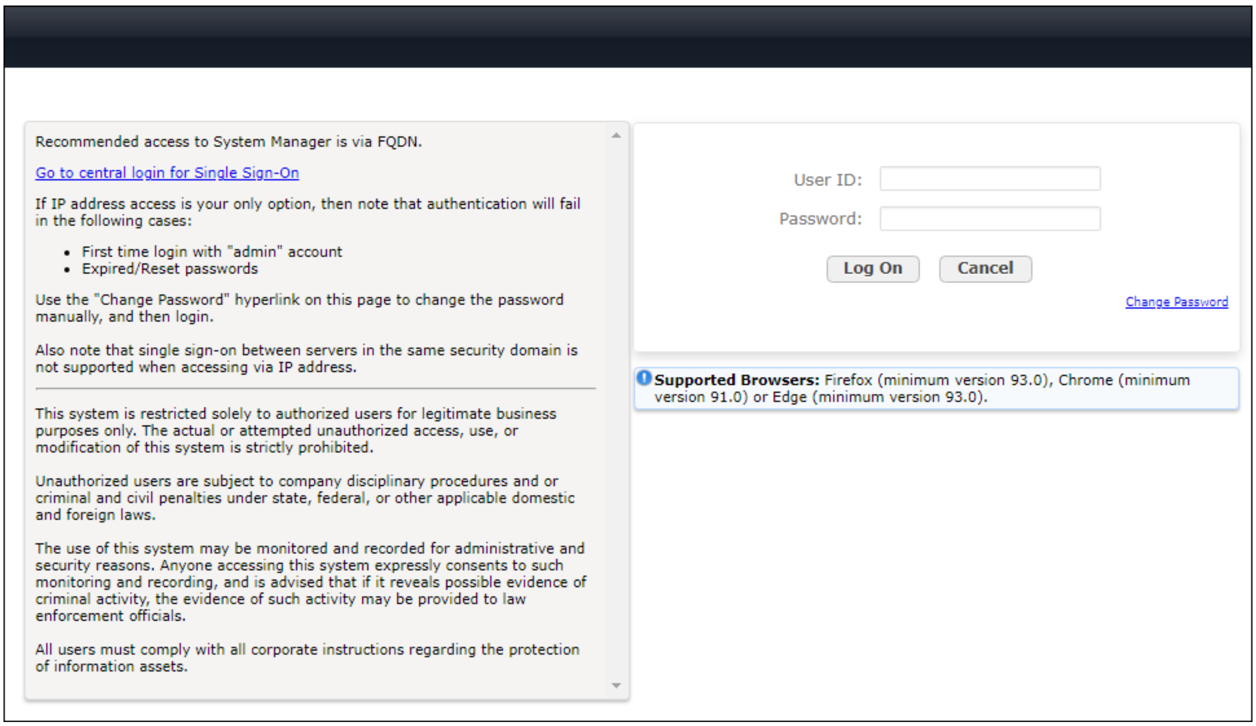
6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The steps include the following areas.

- Launch System Manager
- Administer SIP Users

6.1. Launch System Manager

Access Session Manager Administration web interface by entering **http://<ip-address>/SMGR** in a web browser, where **<ip-address>** is the IP address of System Manager. Log in using the appropriate credentials.



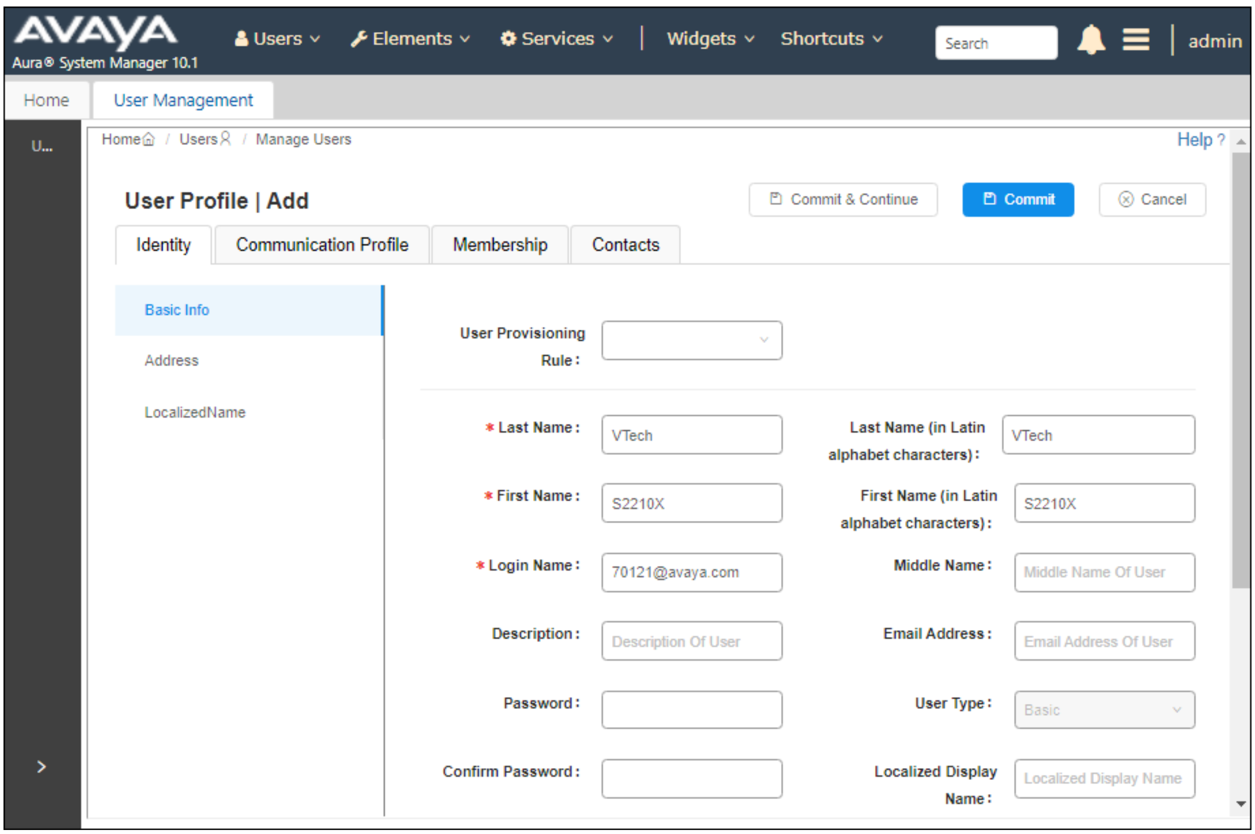
6.2. Administer SIP Users

CTM-S2210-X/CTM-S2211-SPK is administered as a SIP user on Session Manager by the following steps. This configuration is automatically synchronized with Communication Manager. In Session Manager, select **Users →User Management →Manage Users** to display the **User Management** screen (not shown). Click **+ New** to add a user.

6.2.1. Identity

Enter values for the following required attributes for a new SIP user in the **New User Profile** screen:

- **Last Name**: Enter the last name of the user, e.g., *VTech*
- **First Name**: Enter the first name of the user, e.g., *S2210X*
- **Login Name**: Enter <extension>@<sip domain> of the user (e.g., *70121@avaya.com*)

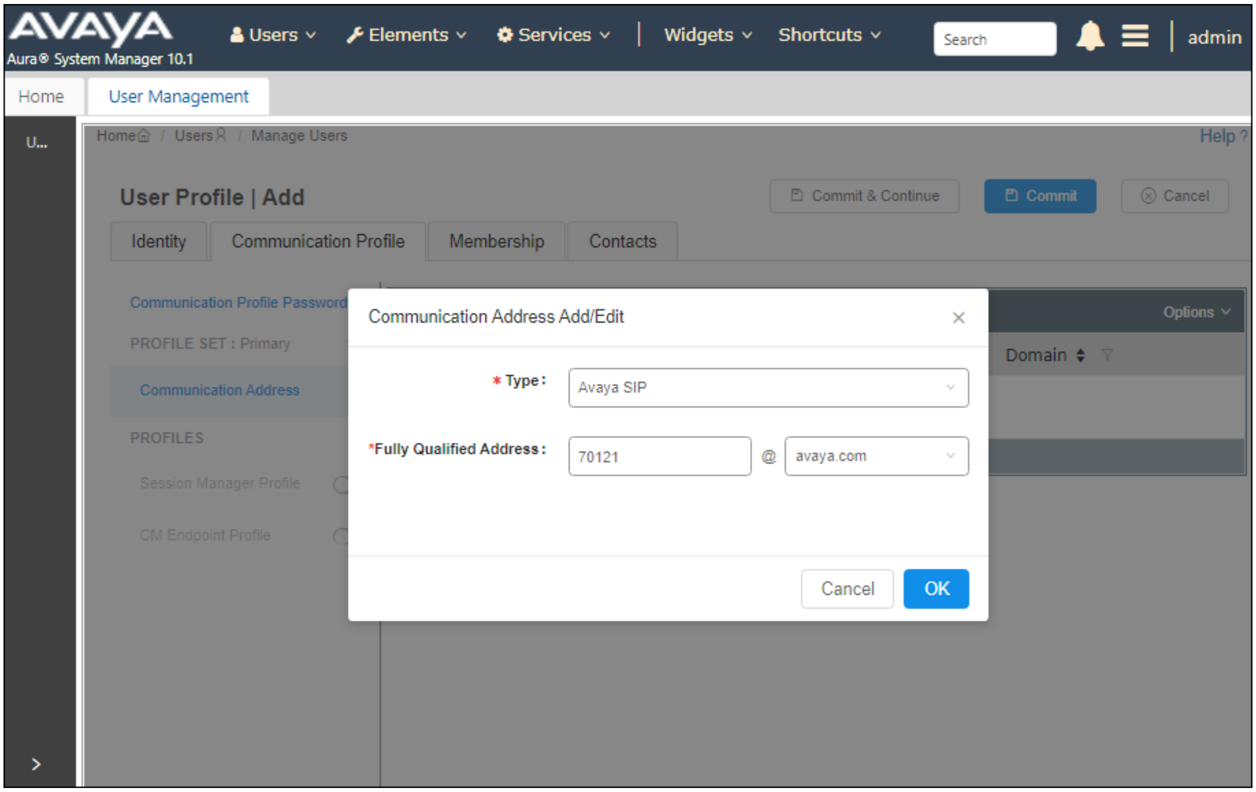


6.2.2. Communication Address

Select the **Communication** Profile tab. Select **Communication Address** in the left hand side list and click **+ New** (not shown).

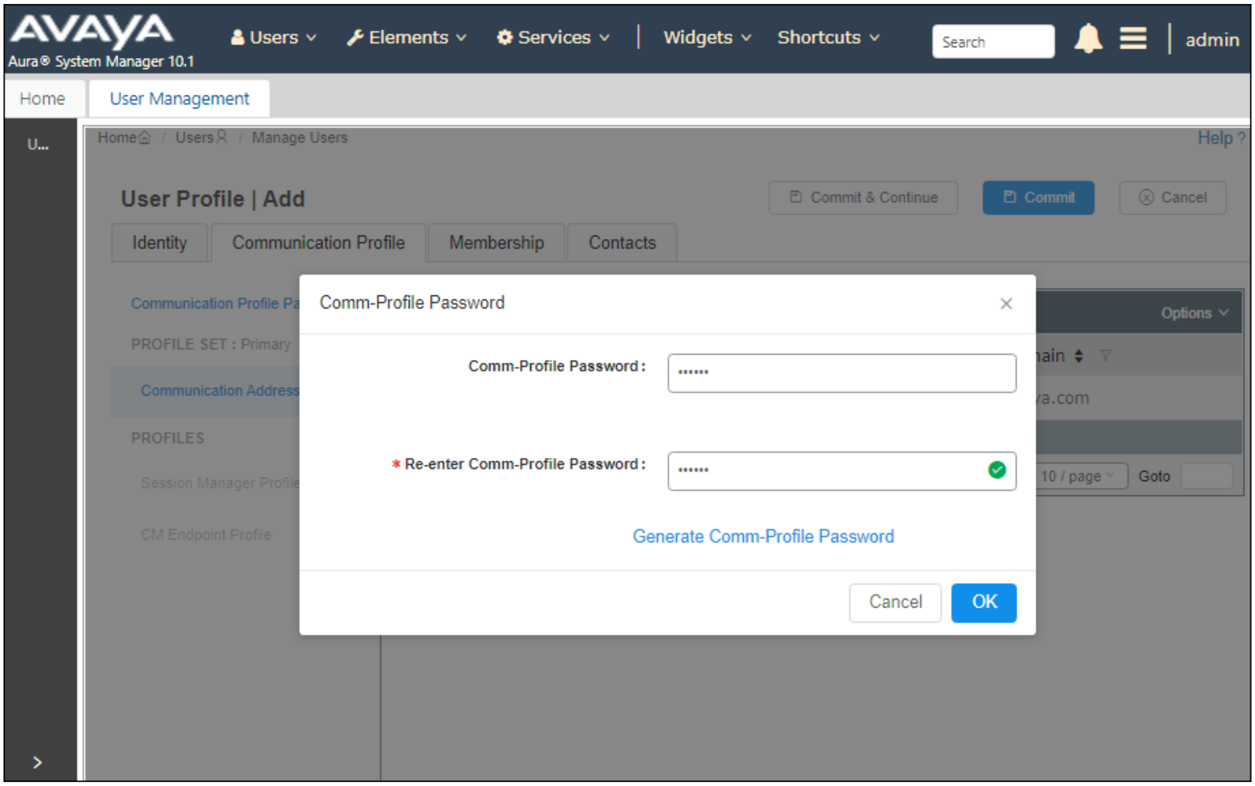
Enter the following attributes for the **Communication Address**:

- **Type:** Select *Avaya SIP* from the drop-down list
- **Fully Qualified Address:** Enter the extension number (e.g.,*70121*)
- **Domain:** Enter the domain (e.g., *avaya.com*)



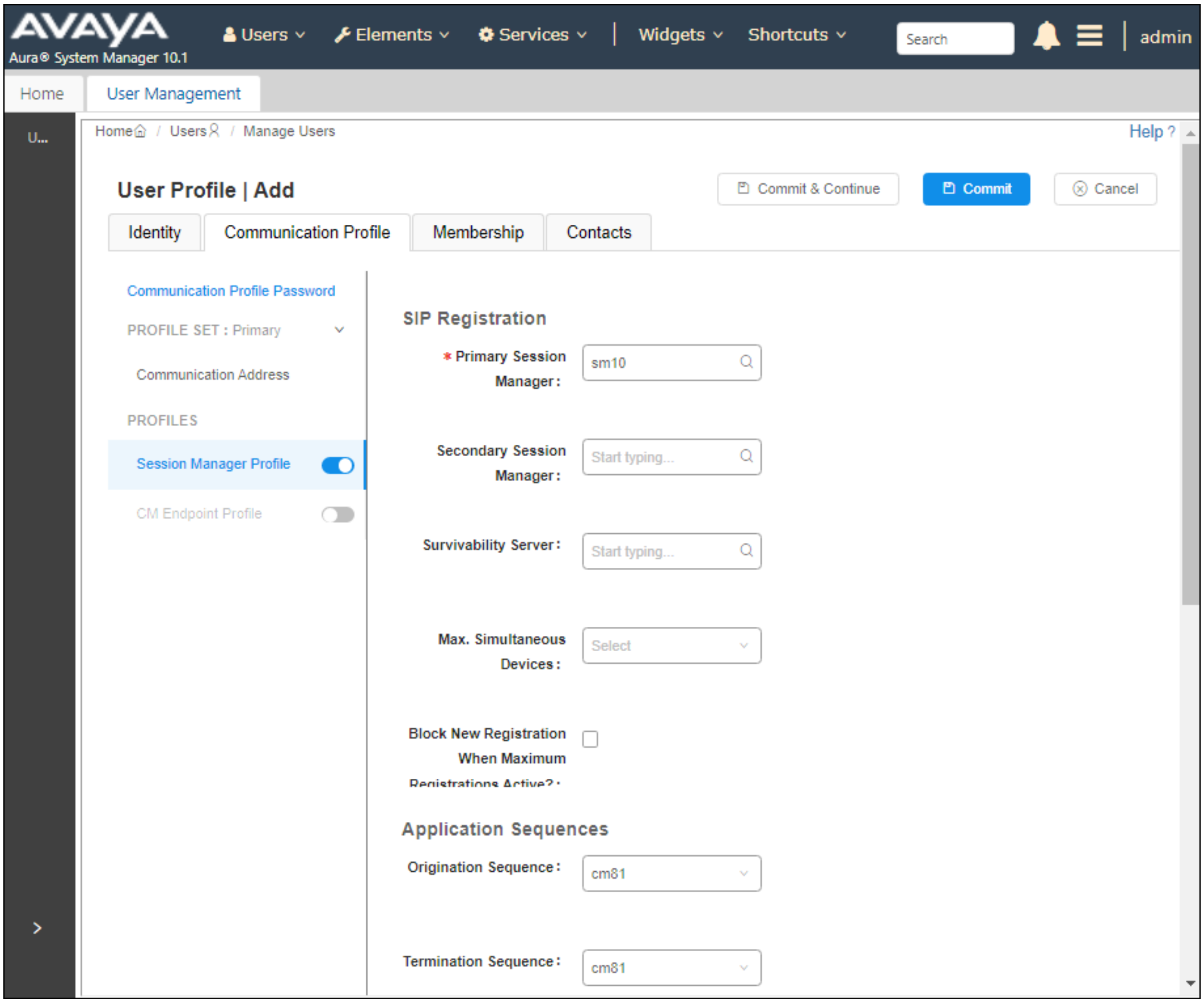
6.2.3. Communication Profile Password

Click the **Communication Profile Password** tab and in the **Comm-Profile Password** and **Re-enter Comm-Profile Password** fields, enter a numeric password. This will be used to register the device. Click **OK**.



6.2.4. Session Manager Profile

Click on the **Session Manager Profile** slide button. For **Primary Session Manager**, **Origination Sequence**, **Termination Sequence**, and **Home Location** (not shown), select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

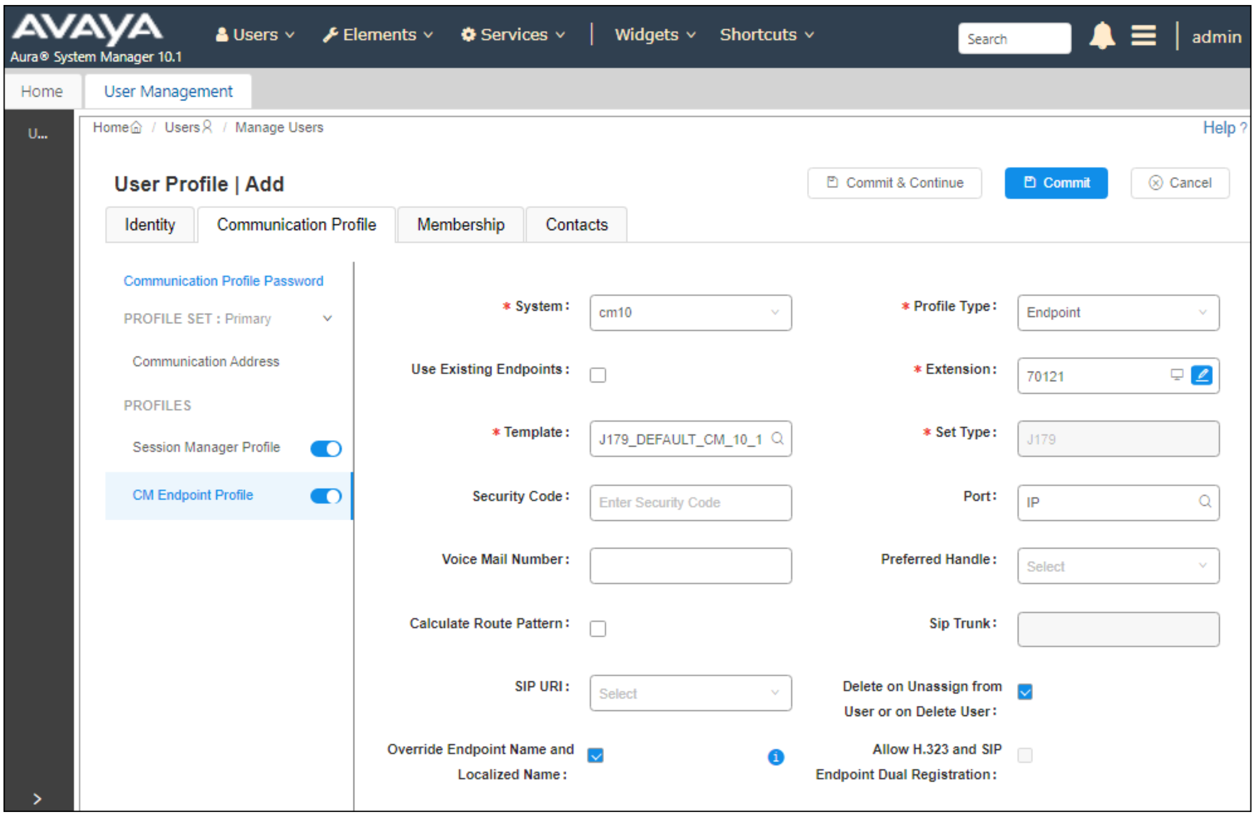


6.2.5. CM Endpoint Profile

Click on the **CM Endpoint Profile** slide button. Fill in the following fields:

- **System**: Select the relevant Communication Manager SIP Entity (e.g., *cm10*)
- **Profile Type**: Select *Endpoint*
- **Template**: Select *J179_DEFAULT_CM_10_1*
- **Extension**: Enter the extension number (e.g., *70121*)

Click on the **Editor** icon in the Extension field to edit Communication Manager settings. Input the appropriate **Coverage Path 1** number (not shown) configured to route unanswered calls to voicemail. Click **Done** to close the Endpoint Editor. Click **Commit**.



7. Configure VTech CTM-S2210-X/CTM-S2211-SPK Corded SIP Hotel Room Phones

The steps to configure CTM-S2210-X/CTM-S2211-SPK to integrate with Communication manager are as follows:

- Configure IP Address
- Launch Web Interface
- Configure SIP Account
- Install CA Certificate
- Modify Codec Settings

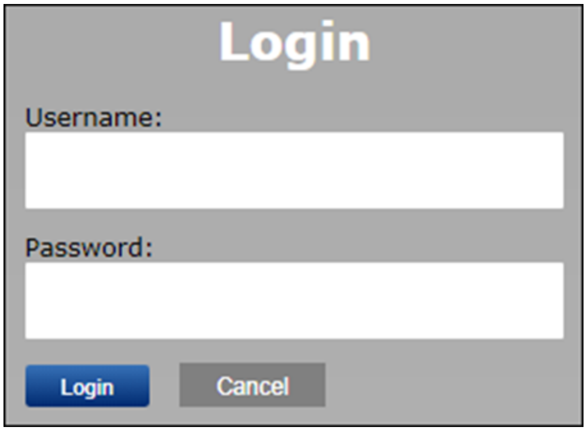
7.1. Configure IP Address

CTM-S2210-X/CTM-S2211-SPK is configured for DHCP as a factory default. The following steps provide network connectivity and determine the phone IP address for use in launching administration detailed in **Section 7.2**:

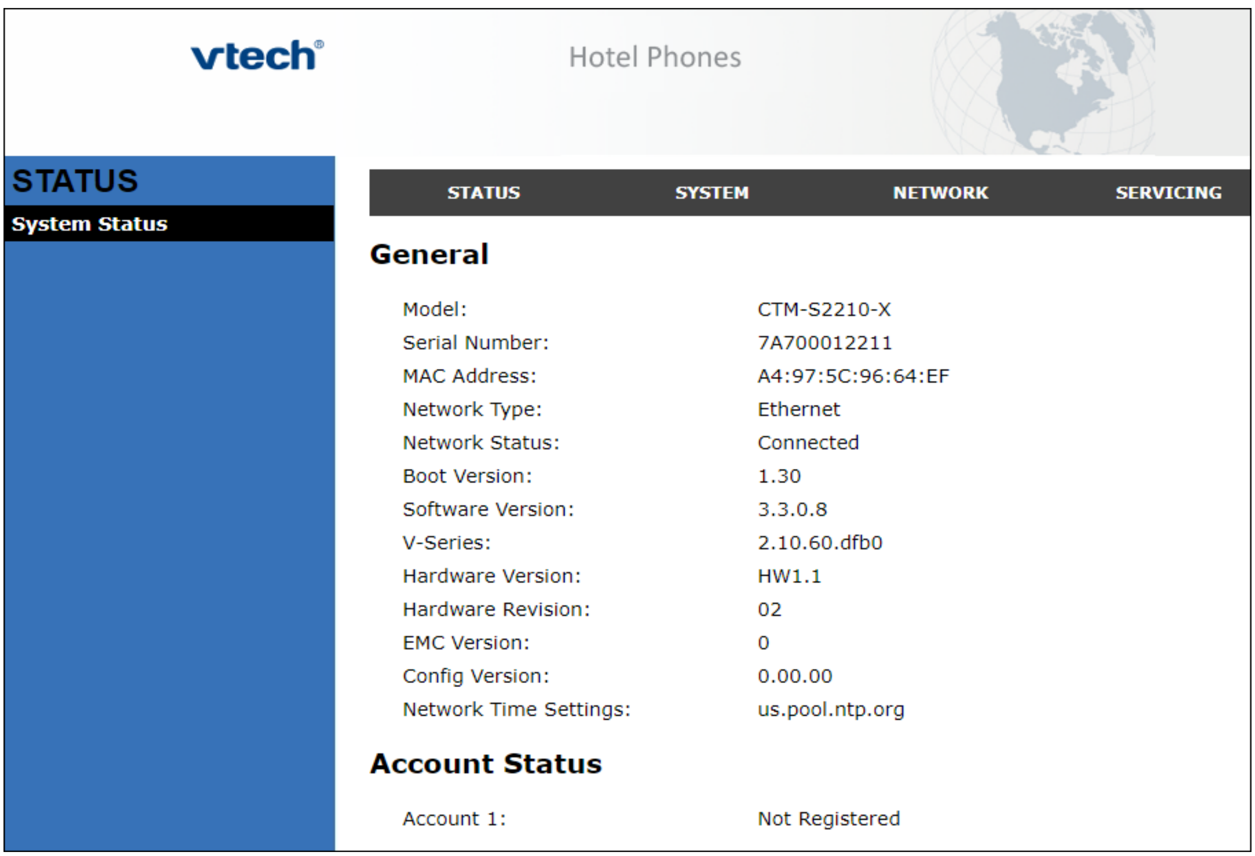
- Connect the LAN port of CTM-S2210-X/CTM-S2211-SPK to a Power over Ethernet (PoE) switch
- Determine the assigned IP address. Use the built-in voice menu which will read out the IP address. The voice menu is accessed by pressing **SPEAKER * * * ***. For more information, refer to CTM-S2210-X/CTM-S2211-SPK user manuals obtained at http://vtechhotelphones.com.

7.2. Launch Web Interface

The phone administration is done though a web interface. To access web administration, invoke the web login page using the **IP address** obtained from **Section 7.1** using the URL **https://<IP address**>. The login prompt is displayed.



Enter the appropriate **Username** and **Password.** Once logged in, the default settings display. The status for CTM-S2210-X is shown.

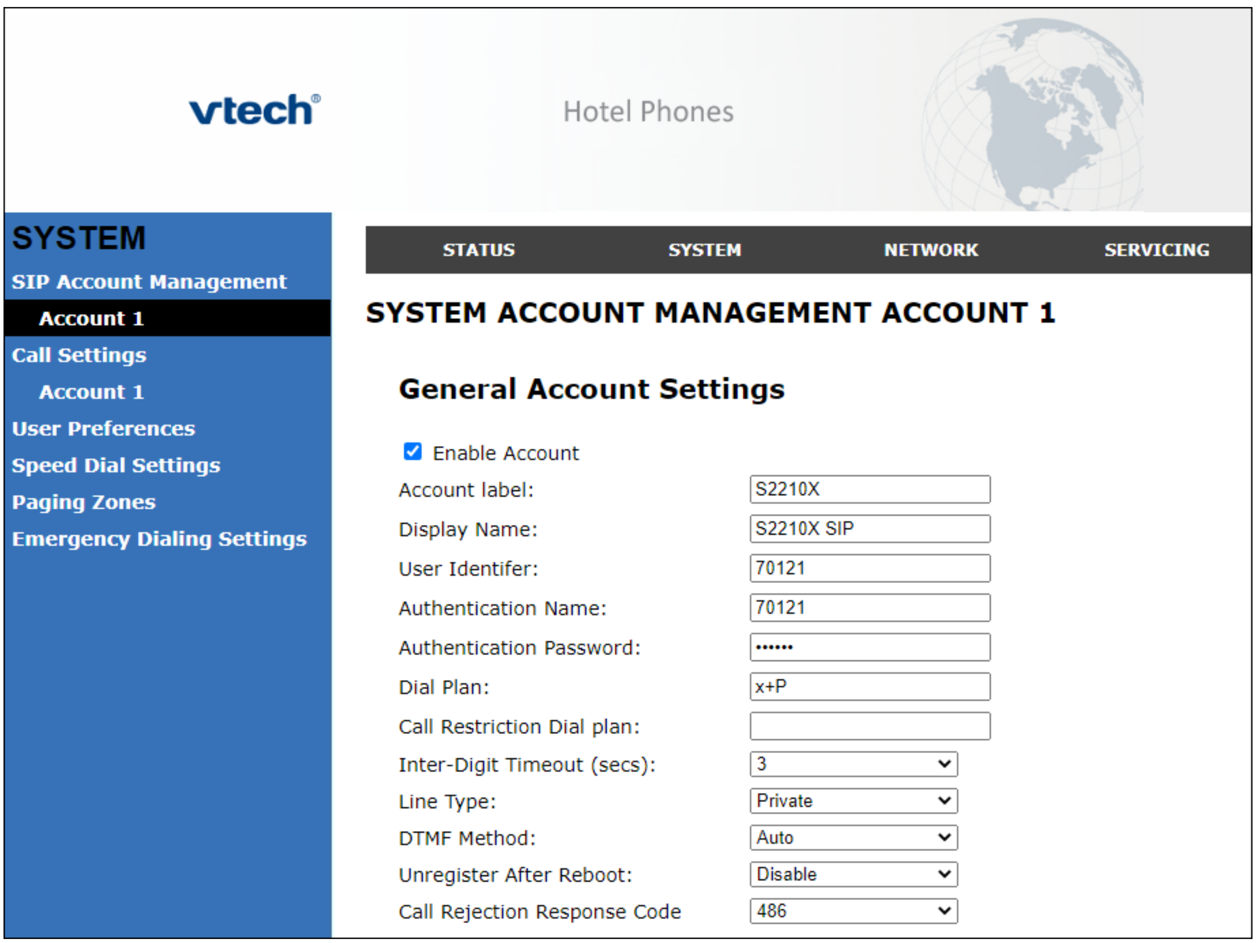


Note: If firmware upgrades are needed, consult the configuration guide for instructions Refer to http://vtechhotelphones.com.

7.3. Configure SIP Account

To register CTM-S2210-X/CTM-S2211-SPK to Session Manager, Select **System** from the toolbar, then **Account 1** from the left-hand side list. Under the **General Account Settings** heading, input the following:

- **Enable Account:** Click the corresponding checkbox
- **Account Label**: A descriptive string (e.g., *S2210X*)
- **Display Name**: The desired display name (e.g., *S2210X SIP*)
- **User Identifier**: An appropriate string (e.g., *70121*)
- **Authentication Name**: Enter the extension number (e.g., *70121*)
- **Authentication Password**: Enter the password

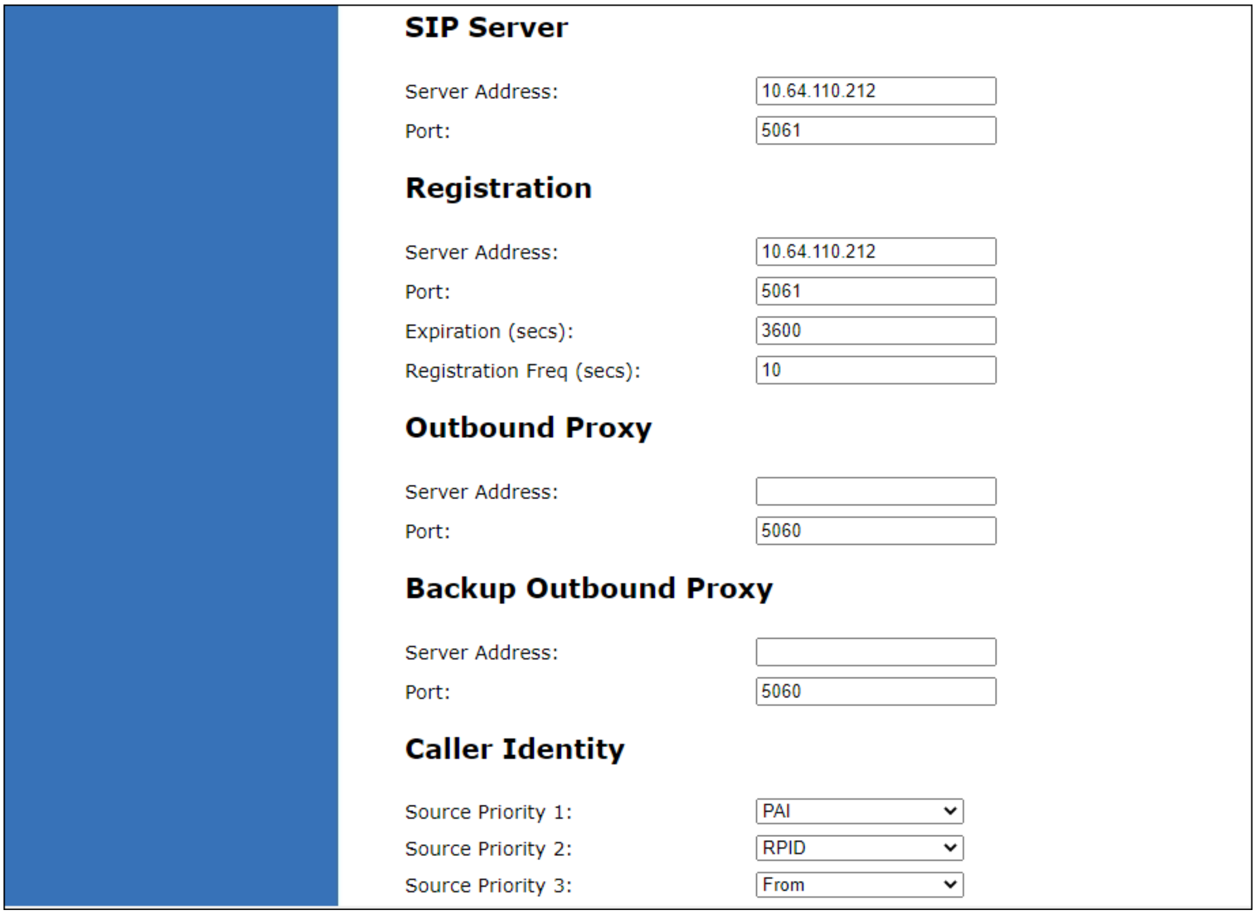


Under the **SIP Server** heading, enter the following:

- **Server Address**: Session Manager IP address (e.g., *10.64.110.212*)
- **Port**: *5061*

Under the **Registration** heading, enter the following:

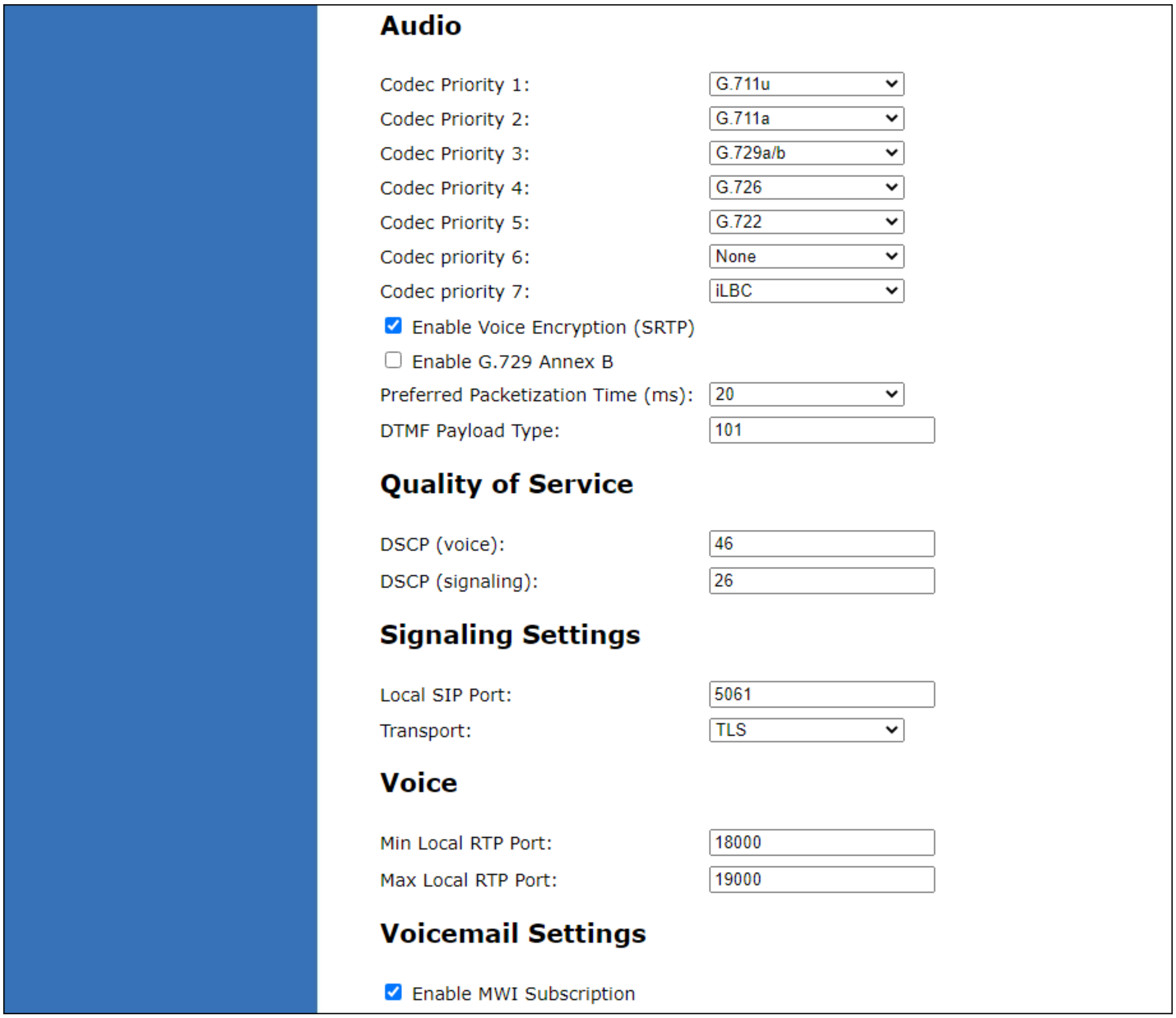
- **Server Address**: Session Manager IP address (e.g., *10.64.110.212*)
- **Port**: *5061*



Continuing on the same page, under the **Audio** heading, select **Enable Voice Encryption (SRTP)**. Under the **Signaling Settings** heading, input the following:

- **Local SIP Port:** *5061*
- **Transport:** *TLS*

Under the **Voicemail Settings** header, select **Enable MWI Subscription.** Click **Save** (not shown).

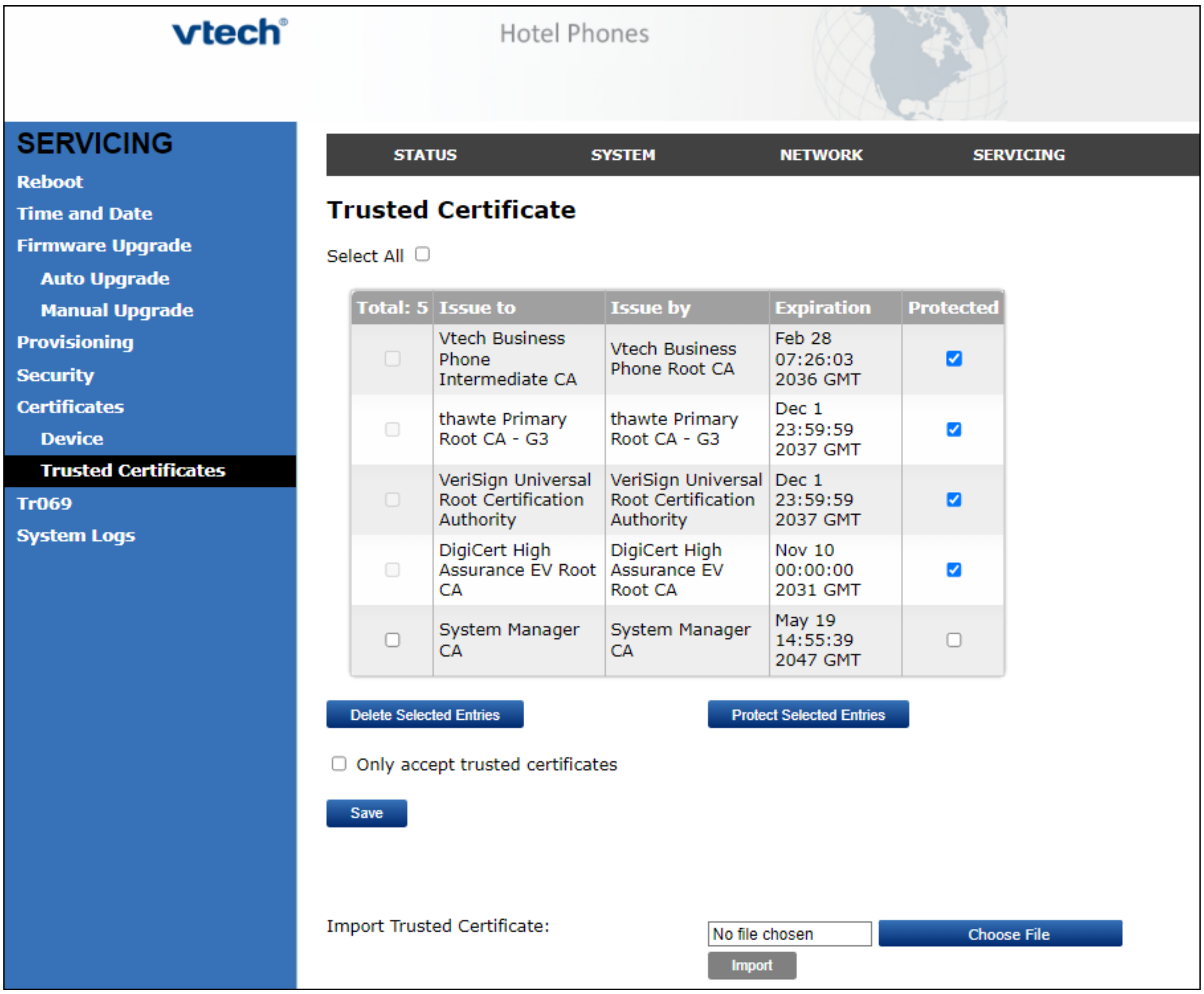


7.4. Install CA Certificate

Note: The Session Manager CA certificate file must be installed in the VTech CTM-S2210-X/CTM-S2211-SPK Trusted Certificate store for validation of the Session Manager identity certificate offered during the TLS handshake.

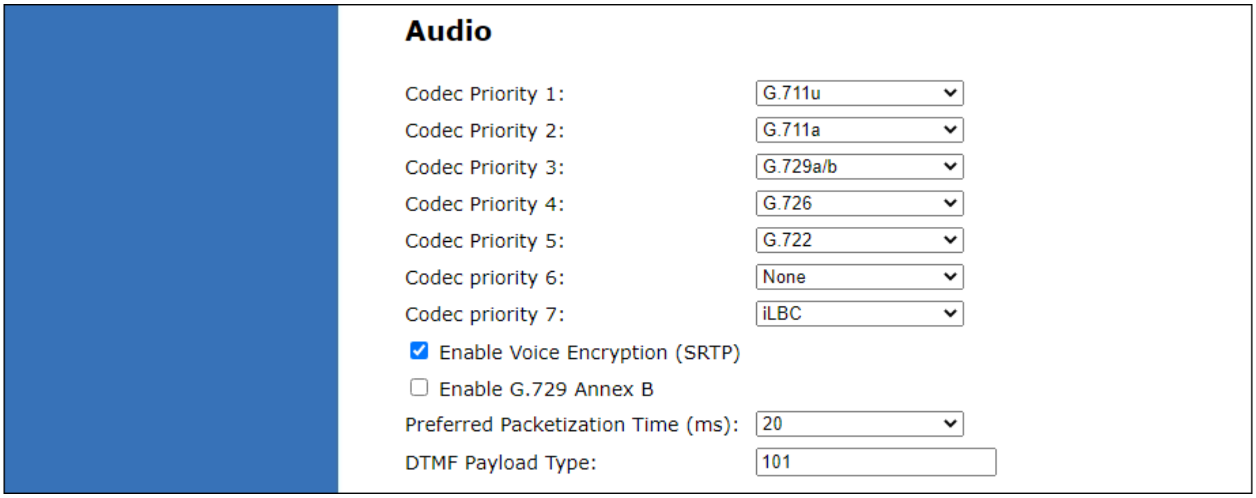
Note: After **Only accept trusted certificates** has been set and saved, subsequent access to the **Trusted Certificates** page will not show that it has been set. The setting can be verified by exporting the phone configuration to a text file in **Provisioning → Export Configuration**. Review the file and verify the entry *provisioning.check_trusted_certificate = 1* exists.

To install the CA certificate, select **SERVICING** from the toolbar, then **Trusted Certificates** from the left-hand side list. Click on **Choose File** and select the CA certificate. Select **Only accept trusted certificates** (not shown). Click **Import** (not shown).The CA should appear in the **Trusted Certificate** list.



7.5. Modify Codec Settings

Modify the codec settings by selecting **SYSTEM** in the toolbar and **Account 1** in the left hand side selections. Under the **Audio** heading, select the desired codecs in priority.



Click **Save**.

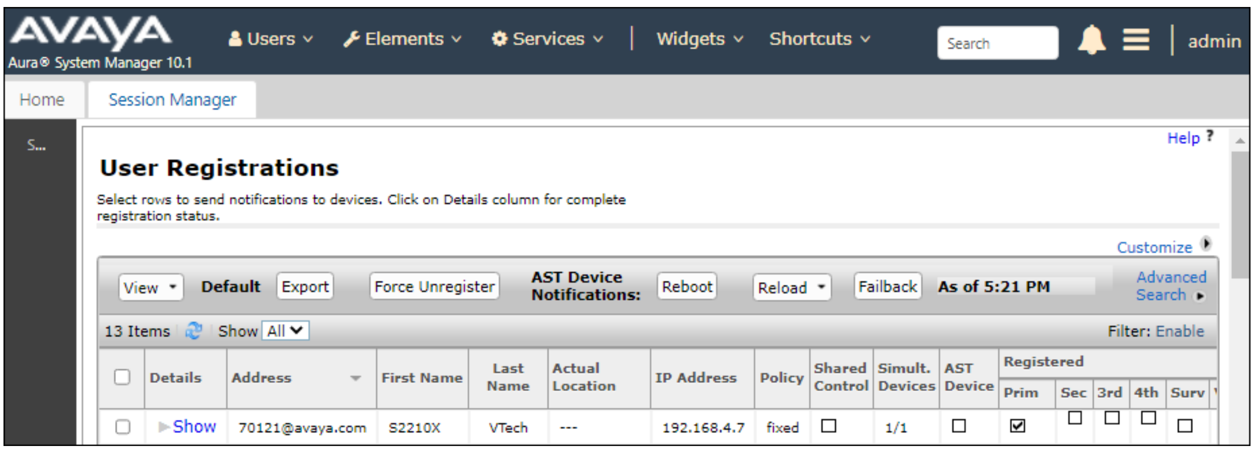
8. Verification Steps

The proper configuration of CTM-S2210-X/CTM-S2211-SPK with Avaya Session Manager and Avaya Communication Manager is verified by the following steps.

8.1. View Session Manager Status

Verify CTM-S2210-X/CTM-S2211-SPK has successfully registered with Session Manager. In System Manager, Navigate to **Elements →Session Manager →System Status →User Registrations. C**heck the registration status by the following:

- Verify CTM-S2210-X (here *70121*) is registered with Session Manager by noting the registered users include *70121*.



8.2. Call Verification

Verify that basic calls can be made from and to CTM-S2210-X/CTM-S2211-SPK and another telephone registered to Session Manager.

9. Conclusion

These Application Notes describe the configuration steps required to integrate VTech CTM-S2101/CTM-S2210-X/CTM-S2212 and CTM-S2211-X/CTM-S2211-SPK/CTM-S2213 Corded SIP Hotel Room Phones with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The VTech CTM-S2101/CTM-S2210-X/CTM-S2212 and CTM-S2211-X/CTM-S2211-SPK/CTM-S2213 Corded SIP Hotel Room Phones register to Avaya Aura® Session Manager. Calls were then established with Avaya H.323 / SIP desk phones and the PSTN. In addition, basic telephony features were verified. All feature and serviceability test cases were completed successfully with observations noted in **Section 2.2**.

10. Additional References

This section references the Avaya documentation relevant to these Application Notes.

Avaya product documentation is available at https://support.avaya.com.

[1] *Avaya Aura® Communication Manager Feature Description and Implementation*, Issue 6, Release 10.1, September 2022.
[2] *Administering Avaya Aura® Session Manager*, Release 10.1, Issue 4, September 2022.
[3] *Administering Avaya Aura® System Manager*, Issue 7, Release 10.1.x, September 2022.

VTech CTM-S2101/CTM-S2210-X/CTM-S2212 and CTM-S2211-X/CTM-S2211-SPK/CTM-S2213 SIP Hotel Phones product documentation is available at https://vtechhotelphones.com.

[4] *SIP Corded Series Master User Guide*, September 1, 2015.

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.

vtech

VTech Technologies Canada Ltd.

Date: October 12, 2022

Declaration of Conformance

We, VTech Technologies Canada LTD., declare under sole responsibility that product series CTM-S2212, CTM-S2210-X, and CTM-S2101 all share the same hardware circuitry, software, SIP stack, and firmware version. Therefore the products are expected to behave in the same manner. Furthermore, these products are a functional superset of the other products in the CTM series. The differences between the different models in the series are detailed in the table below.

Product Name	Model	Description
CTM-S2212	CTM-S2212	Corded SIP Hospitality Room Phone
CTM-S2210-X	CTM-S2210-X	Corded SIP Hospitality Room Phone
CTM-S2101	CTM-S2101	Corded SIP Hospitality Lobby Phone

Please do not hesitate to contact should you require further information.
Thank you,

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VTech Technologies Canada Ltd.

Date: October 12, 2022

Declaration of Conformance

We, VTech Technologies Canada LTD., declare under sole responsibility that product series CTM-S2213 and CTM-S2211-SPK, and CTM-S2211-X all share the same hardware circuitry, software, SIP stack, and firmware version. Therefore the products are expected to behave in the same manner. Furthermore, these products are a functional superset of the other products in the CTM series. The differences between the different models in the series are detailed in the table below.

Product Name	Model	Description
CTM-S2213	CTM-S2213	Corded SIP Hospitality Room Phone without Speakerphone
CTM-S2211-SPK	CTM-S2211-SPK	Corded SIP Hospitality Room Phone with Speakerphone
CTM-S2211-X	CTM-S2211-X	Corded SIP Hospitality Room Phone without Speakerphone

Please do not hesitate to contact should you require further information.
Thank you,

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