



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.

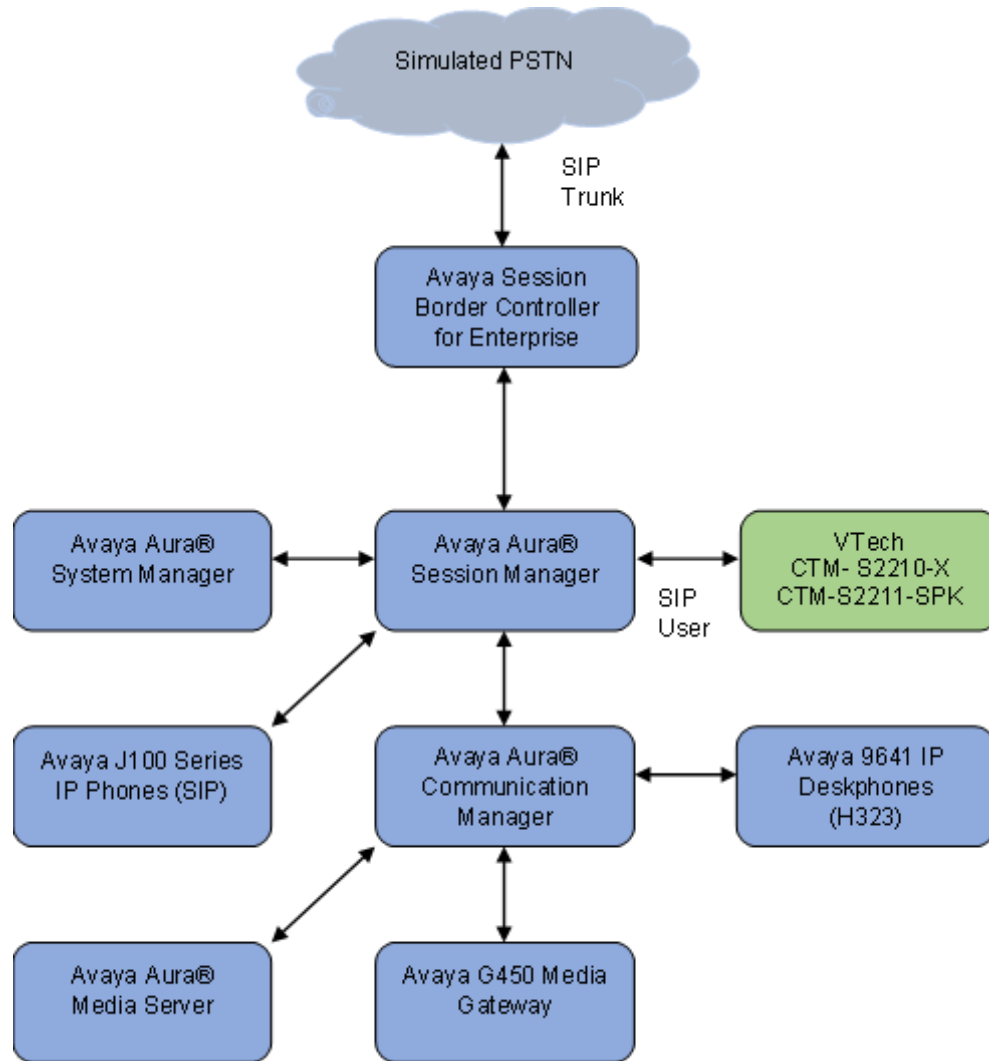


Figure 1: Avaya Test Configuration for CTM-S2210-X/CTM-S2211-SPK Coded SIP Hotel Room Phones

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		
H.323 IP ENDPOINTS		AUDIO RESOURCE RESERVATION PARAMETERS
H.323 Link Bounce Recovery? y		RSVP Enabled? n
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

Page1 of 2

IP MEDIA PARAMETERS

Codec Set: 1

Audio Codec	Silence Suppression	Frames Per Pkt	Packet 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 SRTP: 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.

The screenshot shows the login page for the Avaya Aura Session Manager Administration web interface. The page is divided into two main sections. The left section contains a scrollable area with the following text: "Recommended access to System Manager is via FQDN. [Go to central login for Single Sign-On](#) If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

 Use the "Change Password" hyperlink on this page to change the password manually, and then login. Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address. This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited. Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws. The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials. All users must comply with all corporate instructions regarding the protection of information assets."

The right section contains the login form with the following elements: "User ID:" followed by a text input field, "Password:" followed by a text input field, "Log On" and "Cancel" buttons, and a "Change Password" hyperlink. Below the login form is a blue box with the text: "Supported Browsers: Firefox (minimum version 93.0), Chrome (minimum version 91.0) or Edge (minimum version 93.0)."

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*)

The screenshot shows the Avaya Aura System Manager 10.1 interface. The top navigation bar includes the Avaya logo, a search bar, and a user profile icon labeled 'admin'. The main navigation menu on the left includes 'Home', 'User Management', and 'U...'. The 'User Management' section is expanded, showing 'Home / Users / Manage Users'. The 'User Profile | Add' form is displayed, with tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Identity' tab is selected, showing a 'Basic Info' section. The form includes fields for 'User Provisioning Rule', 'Last Name', 'First Name', 'Login Name', 'Description', 'Password', 'Confirm Password', 'Last Name (in Latin alphabet characters)', 'First Name (in Latin alphabet characters)', 'Middle Name', 'Email Address', 'User Type', and 'Localized Display Name'. The fields are populated with example values: 'Last Name' is 'VTech', 'First Name' is 'S2210X', 'Login Name' is '70121@avaya.com', 'Description' is 'Description Of User', 'Password' is empty, 'Confirm Password' is empty, 'Last Name (in Latin alphabet characters)' is 'VTech', 'First Name (in Latin alphabet characters)' is 'S2210X', 'Middle Name' is 'Middle Name Of User', 'Email Address' is 'Email Address Of User', 'User Type' is 'Basic', and 'Localized Display Name' is 'Localized Display Name'. Buttons for 'Commit & Continue', 'Commit', and 'Cancel' are visible at the top right of the form.

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*)

The screenshot displays the Avaya Aura System Manager 10.1 web interface. The top navigation bar includes the Avaya logo, a search bar, and user information (admin). The main content area is titled 'User Profile | Add' and features tabs for Identity, Communication Profile, Membership, and Contacts. The 'Communication Profile' tab is active, and the 'Communication Address' section is highlighted in the left sidebar. A modal dialog box titled 'Communication Address Add/Edit' is open in the foreground. It contains two fields: '* Type:' with a dropdown menu set to 'Avaya SIP', and '* Fully Qualified Address:' with a text input containing '70121' and a domain dropdown set to 'avaya.com'. The dialog has 'Cancel' and 'OK' buttons at the bottom.

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**.

The screenshot displays the Avaya Aura System Manager 10.1 interface. The top navigation bar includes the Avaya logo, navigation links (Users, Elements, Services, Widgets, Shortcuts), a search bar, and a user profile (admin). The main content area shows the 'User Profile | Add' dialog with tabs for Identity, Communication Profile, Membership, and Contacts. The 'Communication Profile' tab is selected, showing fields for 'Communication Profile Password' and 'Re-enter Comm-Profile Password'. A modal window titled 'Comm-Profile Password' is open, prompting for a password and its re-entry. The modal window has a 'Generate Comm-Profile Password' link and 'Cancel' and 'OK' buttons.

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.

The screenshot displays the Avaya Aura System Manager 10.1 interface. The top navigation bar includes the Avaya logo, a search bar, and user roles (Users, Elements, Services, Widgets, Shortcuts). The main header shows 'Home' and 'User Management'. The breadcrumb trail indicates 'Home / Users / Manage Users'. The 'User Profile | Add' form is open, with tabs for Identity, Communication Profile, Membership, and Contacts. The 'Communication Profile' tab is active, showing a 'Communication Profile Password' section with a dropdown for 'PROFILE SET : Primary' and a 'Communication Address' field. Below this, the 'PROFILES' section has a 'Session Manager Profile' toggle switch turned on and a 'CM Endpoint Profile' toggle switch turned off. The 'SIP Registration' section contains fields for 'Primary Session Manager' (set to 'sm10'), 'Secondary Session Manager' (set to 'Start typing...'), 'Survivability Server' (set to 'Start typing...'), 'Max. Simultaneous Devices' (set to 'Select'), and a checkbox for 'Block New Registration When Maximum Registrations Active?'. The 'Application Sequences' section has dropdowns for 'Origination Sequence' (set to 'cm81') and 'Termination Sequence' (set to 'cm81'). Action buttons 'Commit & Continue', 'Commit', and 'Cancel' are at the top right of the form.

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**.

The screenshot shows the Avaya Aura System Manager 10.1 interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 10.1', and tabs for Users, Elements, Services, Widgets, and Shortcuts. A search bar and user profile 'admin' are on the right. The main content area is titled 'User Profile | Add' and has tabs for Identity, Communication Profile (selected), Membership, and Contacts. On the left, a sidebar shows 'Communication Profile Password', 'PROFILE SET : Primary', 'Communication Address', 'PROFILES', 'Session Manager Profile' (disabled), and 'CM Endpoint Profile' (enabled). The main form fields are as follows:

Field	Value
* System :	cm10
* Profile Type :	Endpoint
* Extension :	70121
* Template :	J179_DEFAULT_CM_10_1
* Set Type :	J179
Security Code :	Enter Security Code
Port :	IP
Voice Mail Number :	
Preferred Handle :	Select
Sip Trunk :	
SIP URI :	Select
Calculate Route Pattern :	<input type="checkbox"/>
Override Endpoint Name and Localized Name :	<input checked="" type="checkbox"/>
Delete on Unassign from User or on Delete User :	<input checked="" type="checkbox"/>
Allow H.323 and SIP Endpoint Dual Registration :	<input type="checkbox"/>

Buttons at the top right: Commit & Continue, Commit, Cancel.

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

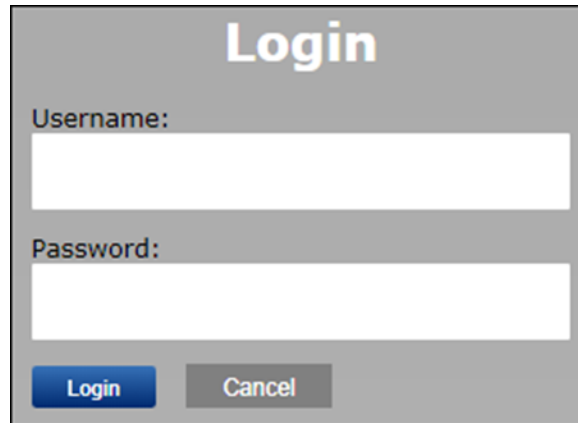
7.1. Configure IP Address

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**:

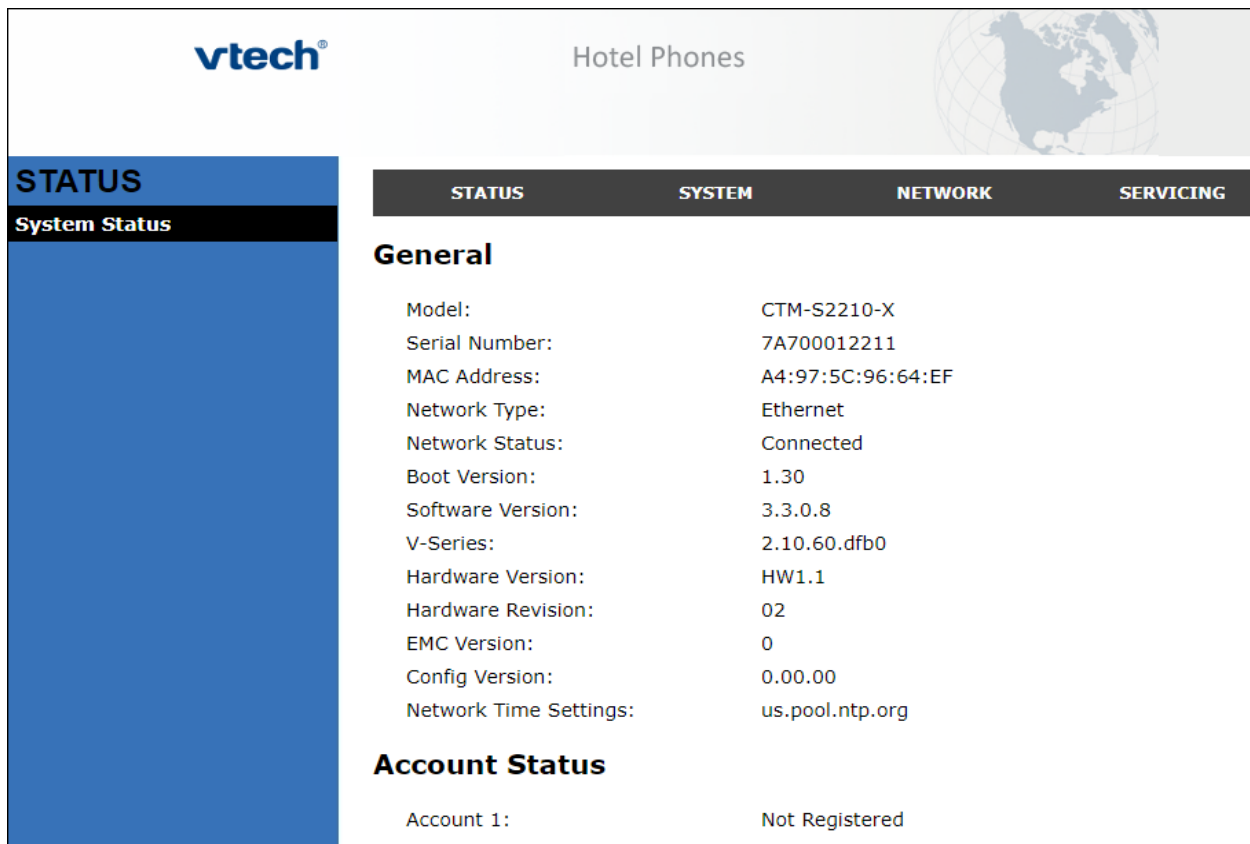
- Connect the LAN port of CTM-S2210-X/CTM-S2211-SPK to a Power over Ethernet (PoE) switch
- Determine the assigned IP address. Use the built-in voice menu which will read out the IP address. The voice menu is accessed by pressing **SPEAKER * * * ***. For more information, refer to CTM-S2210-X/CTM-S2211-SPK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

7.2. Launch Web Interface

The phone administration is done through 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.

A login dialog box with a gray background. At the top, the word "Login" is written in a large, bold, white font. Below it, there are two input fields. The first is labeled "Username:" in a small, bold, gray font. The second is labeled "Password:" in a small, bold, gray font. At the bottom, there are two buttons: a blue "Login" button and a gray "Cancel" button.

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

The Vtech Hotel Phones web interface. The top header features the Vtech logo on the left, "Hotel Phones" in the center, and a globe icon on the right. Below the header, there is a navigation bar with four tabs: "STATUS", "SYSTEM", "NETWORK", and "SERVICING". The "STATUS" tab is selected and highlighted in blue. Under the "STATUS" tab, there is a sub-section titled "System Status". The main content area displays the "General" status for the phone. It lists various system parameters and their values. Below the "General" section, there is an "Account Status" section showing the account registration status.

General	
Model:	CTM-S2210-X
Serial Number:	7A700012211
MAC Address:	A4:97:5C:96:64:EF
Network Type:	Ethernet
Network Status:	Connected
Boot Version:	1.30
Software Version:	3.3.0.8
V-Series:	2.10.60.dfb0
Hardware Version:	HW1.1
Hardware Revision:	02
EMC Version:	0
Config Version:	0.00.00
Network Time Settings:	us.pool.ntp.org

Account Status	
Account 1:	Not Registered

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

The screenshot displays the Vtech Hotel Phones configuration web interface. At the top, the Vtech logo and 'Hotel Phones' text are visible, along with a globe icon. A navigation bar includes 'STATUS', 'SYSTEM', 'NETWORK', and 'SERVICING'. The left sidebar shows a tree structure with 'SYSTEM' selected, containing 'SIP Account Management', 'Account 1', 'Call Settings', 'Account 1', 'User Preferences', 'Speed Dial Settings', 'Paging Zones', and 'Emergency Dialing Settings'. The main content area is titled 'SYSTEM ACCOUNT MANAGEMENT ACCOUNT 1' and features a 'General Account Settings' section. This section includes a checked 'Enable Account' checkbox and various input fields for account configuration.

Field	Value
Enable Account	<input checked="" type="checkbox"/>
Account label:	S2210X
Display Name:	S2210X SIP
User Identifier:	70121
Authentication Name:	70121
Authentication Password:
Dial Plan:	x+P
Call Restriction Dial plan:	
Inter-Digit Timeout (secs):	3
Line Type:	Private
DTMF Method:	Auto
Unregister After Reboot:	Disable
Call Rejection Response Code	486

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*

	SIP Server	
	Server Address:	<input type="text" value="10.64.110.212"/>
	Port:	<input type="text" value="5061"/>
	Registration	
	Server Address:	<input type="text" value="10.64.110.212"/>
	Port:	<input type="text" value="5061"/>
	Expiration (secs):	<input type="text" value="3600"/>
	Registration Freq (secs):	<input type="text" value="10"/>
	Outbound Proxy	
	Server Address:	<input type="text"/>
	Port:	<input type="text" value="5060"/>
	Backup Outbound Proxy	
	Server Address:	<input type="text"/>
	Port:	<input type="text" value="5060"/>
	Caller Identity	
Source Priority 1:	<input type="text" value="PAI"/>	
Source Priority 2:	<input type="text" value="RPID"/>	
Source Priority 3:	<input type="text" value="From"/>	

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).

Audio	
Codec Priority 1:	<input type="text" value="G.711u"/>
Codec Priority 2:	<input type="text" value="G.711a"/>
Codec Priority 3:	<input type="text" value="G.729a/b"/>
Codec Priority 4:	<input type="text" value="G.726"/>
Codec Priority 5:	<input type="text" value="G.722"/>
Codec priority 6:	<input type="text" value="None"/>
Codec priority 7:	<input type="text" value="iLBC"/>
<input checked="" type="checkbox"/> Enable Voice Encryption (SRTP)	
<input type="checkbox"/> Enable G.729 Annex B	
Preferred Packetization Time (ms):	<input type="text" value="20"/>
DTMF Payload Type:	<input type="text" value="101"/>
Quality of Service	
DSCP (voice):	<input type="text" value="46"/>
DSCP (signaling):	<input type="text" value="26"/>
Signaling Settings	
Local SIP Port:	<input type="text" value="5061"/>
Transport:	<input type="text" value="TLS"/>
Voice	
Min Local RTP Port:	<input type="text" value="18000"/>
Max Local RTP Port:	<input type="text" value="19000"/>
Voicemail Settings	
<input checked="" type="checkbox"/> Enable MWI Subscription	

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.

vtch®

Hotel Phones

SERVICING

Reboot

Time and Date

Firmware Upgrade

Auto Upgrade

Manual Upgrade

Provisioning

Security

Certificates

Device

Trusted Certificates

Tr069

System Logs

STATUS

SYSTEM

NETWORK

SERVICING

Trusted Certificate

Select All ☐

Total: 5

	Issue to	Issue by	Expiration	Protected
<input type="checkbox"/>	Vtech Business Phone Intermediate CA	Vtech Business Phone Root CA	Feb 28 07:26:03 2036 GMT	<input checked="" type="checkbox"/>
<input type="checkbox"/>	thawte Primary Root CA - G3	thawte Primary Root CA - G3	Dec 1 23:59:59 2037 GMT	<input checked="" type="checkbox"/>
<input type="checkbox"/>	VeriSign Universal Root Certification Authority	VeriSign Universal Root Certification Authority	Dec 1 23:59:59 2037 GMT	<input checked="" type="checkbox"/>
<input type="checkbox"/>	DigiCert High Assurance EV Root CA	DigiCert High Assurance EV Root CA	Nov 10 00:00:00 2031 GMT	<input checked="" type="checkbox"/>
<input type="checkbox"/>	System Manager CA	System Manager CA	May 19 14:55:39 2047 GMT	<input type="checkbox"/>

Delete Selected Entries

Protect Selected Entries

☐ Only accept trusted certificates

Save

Import Trusted Certificate:

No file chosen

Choose File

Import

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.

Audio

Codec Priority 1:

Codec Priority 2:

Codec Priority 3:

Codec Priority 4:

Codec Priority 5:

Codec priority 6:

Codec priority 7:

☒ Enable Voice Encryption (SRTP)

☐ Enable G.729 Annex B

Preferred Packetization Time (ms):

DTMF Payload Type:

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**. Check the registration status by the following:

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

AVAYA
Aura® System Manager 10.1

Users ▾ Elements ▾ Services ▾ Widgets ▾ Shortcuts ▾ Search [] admin

Home Session Manager

S...

Help ?

User Registrations

Select rows to send notifications to devices. Click on Details column for complete registration status.

Customize ▾

View ▾ Default Export Force Unregister AST Device Notifications: Reboot Reload ▾ Failback As of 5:21 PM Advanced Search ▾

13 Items Show All ▾ Filter: Enable

	Details	Address	First Name	Last Name	Actual Location	IP Address	Policy	Shared Control	Simult. Devices	AST Device	Registered				
											Prim	Sec	3rd	4th	Surv
<input type="checkbox"/>	Show	70121@avaya.com	S2210X	VTech	---	192.168.4.7	fixed	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

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.

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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 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,

A handwritten signature in black ink, appearing to read "R. Tischler".

Ralph Tischler
Director of Engineering
VTech Technologies Canada LTD.
604-273-5131
ralphtischler@vtech.ca



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,

A handwritten signature in black ink, appearing to read "R. Tischler".

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