



Avaya Solution & Interoperability Test Lab

Application Notes for VTech CTM-S2210-X and CTM-S2211-SPK Corded SIP Hospitality Room Phones with Avaya IP Office 11.1 – 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 v.3.3.0.8 and v3.4.0.0-9, respectively, to interoperate with Avaya IP Office Server Edition 11.1 and Avaya IP Office 500 V2 Expansion System 11.1. VTech CTM-S2101/CTM-S2210-X/CTM-S2212 and CTM-S2211-X/CTM-S2211-SPK/CTM-S2213 Corded SIP Hotel Room Phones register directly with Avaya IP Office 11.1.

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 IP Office Server Edition 11.1 and Avaya IP Office 500 V2 Expansion System 11.1. 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 IP Office Server Edition 11.1 or Avaya IP Office 500 V2 Expansion System 11.1 as a SIP endpoint. Compliance testing used the VTech CTM-S2210-X/CTM-S2211-SPK Corded SIP Hotel Room Phones as representative models. CTM-S2210-X/CTM-S2211-SPK models are corded, one line SIP endpoints. See **Attachment 1** which provides details of VTech CTM-S2210-X Corded SIP Hotel Room Phone equivalency to the CTM-S2101 and CTM-S2212 Corded SIP Hotel Room Phone models. See **Attachment 2** which provide details of VTech CTM-S2211-SPK SIP Hotel 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 to PSTN, Avaya SIP, and Avaya H.323 endpoints 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 VTech CTM-S2210-X/CTM-S2211-SPK utilized enabled capabilities of TLS/SRTP.

2.1. Interoperability Compliance Testing

The following features and functionality were evaluated in the interoperability compliance test:

- Registration of CTM-S2210-X/CTM-S2211-SPK to IP Office.
- Basic call features: Answer, Hold/Resume, Mute/Un-mute, Drop, Message Waiting Indicator, DTMF, Call Waiting, Call Forward.
- Codec negotiation and Direct Media Path
- Hospitality features: Automatic Wakeup Call and Do Not Disturb
- Serviceability testing to validate recovery from network connectivity loss.

2.2. Test Results

All test cases passed with the following observations:

- CTM-S2210-X/CTM-S2211-SPK do not support the following features
 - Call Park/Unpark
 - Call Pickup
 - Hold Timeout
 - Transfer
 - Conference
 - CTM-S2210-X/CTM-S2211-SPK programmable buttons do not support short codes requiring secondary input.
- CTM-S2210-X/CTM-S2211-SPK do not support SDP negotiation capabilities per (RFC5939) between SRTP and non-SRTP modes. Media Security for the associated extensions should be set to Enforced.
- The web administration setting **Only accept trusted certificates** must be set for CTM-S2210-X/CTM-S2211-SPK to validate IP Office'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 Hotel 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 IP Office Server Edition and Avaya IP Office 500 V2 Expansion System.

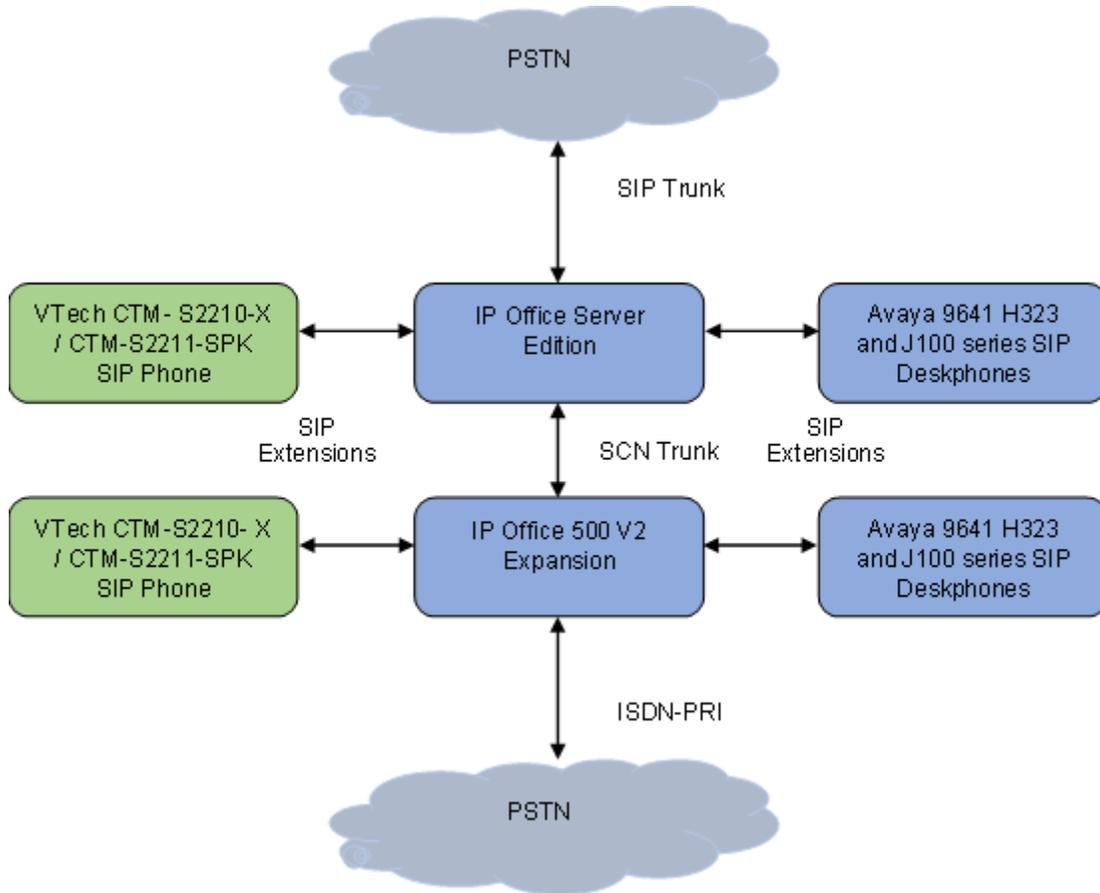


Figure 1: Avaya Interoperability Test Configuration for VTech CTM-S2210-X/CTM-S2211-SPK Corded SIP Hotel Room Phones

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office 500 V2 Expansion	11.1.2.2
Avaya IP Office Server Edition running on Virtual Machine	11.1.2.2
Avaya 9641G IP Deskphones	6.8502 (H.323)
Avaya J129 IP Phone	4.0.7.0.7 (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

Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations.

5. Configure Avaya IP Office Server Edition

This section provides the procedures for configuring Avaya IP Office Server Edition. The procedures include the following areas:

- Verify License
- Obtain LAN IP address
- Administer SIP registrar
- Administer SIP extension for CTM-S2210-X/CTM-S2211-SPK
- Administer SIP user for CTM-S2210-X/CTM-S2211-SPK

Note: This section covers the configuration of Avaya IP Office Server Edition, but the configuration is the same for Avaya IP Office 500 V2 Expansion System.

5.1. Verify License

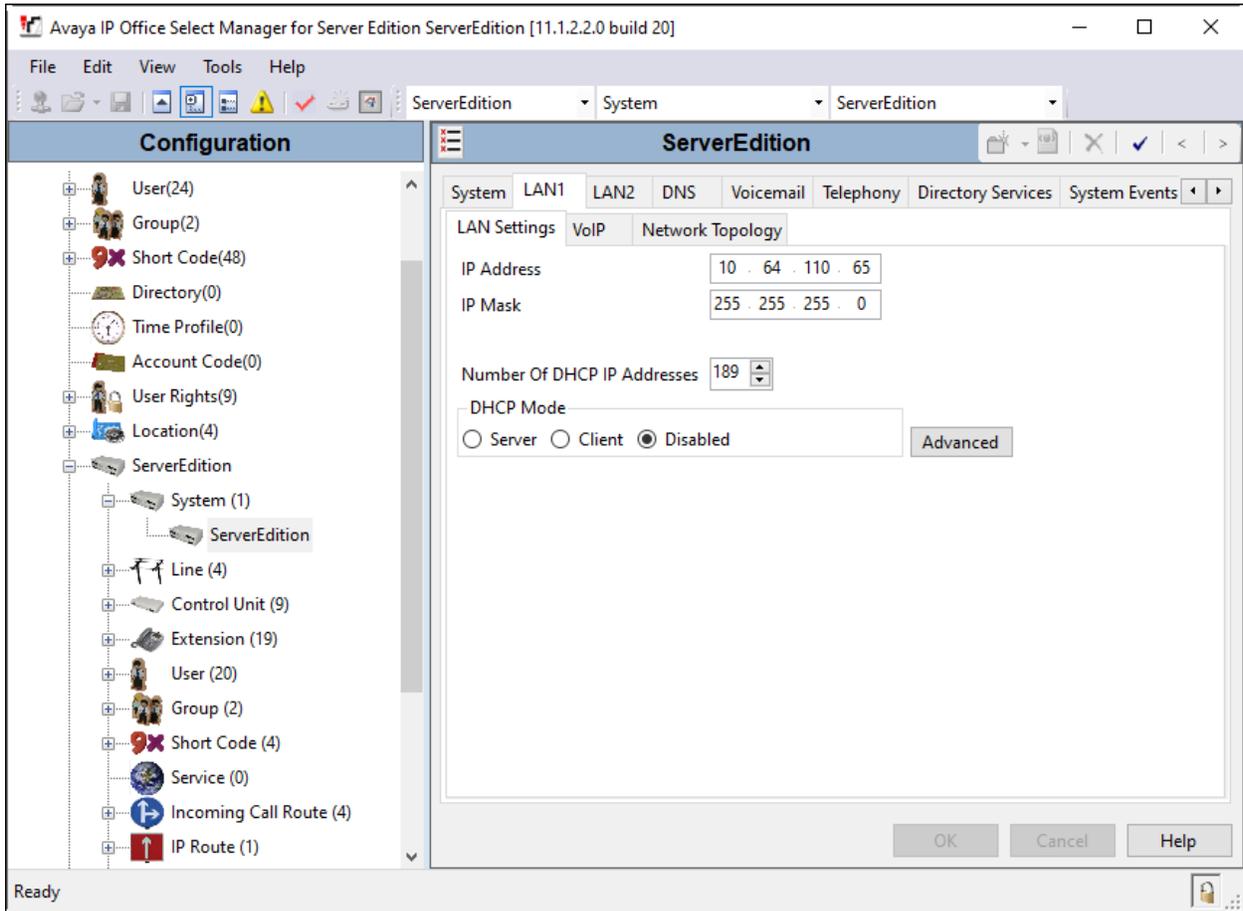
From a PC running the **IP Office Admin Suite**, invoke **IP Office Manager**. Select the proper primary IP Office system, and log in using the appropriate credentials. The Avaya IP Office Manager for Server Edition screen is displayed.

From the configuration tree in the left pane, select **License** under the IP Office system that will be used to display a list of licenses in the right pane. Verify that there are sufficient licenses for **3rd Party IP Endpoints** as shown below.

Feature	Instances	Status	Expiration Date
3rd Party IP Endpoints	1000	Valid	Never
Additional Voicemail Pro Ports	252	Valid	Never
Avaya Contact Center Select	1	Valid	Never
Avaya IP endpoints	1000	Valid	Never
Avaya Mac Softphone	1000	Valid	Never
Avaya Softphone Licence	1000	Valid	Never
Basic User	1000	Obsolete	Never
CTI Link Pro	1	Valid	Never
Devlink3 External Recorder	1	Valid	Never
IP500 Universal PRI (Additional cha...	100	Obsolete	Never
IPSec Tunnelling	1	Obsolete	Never

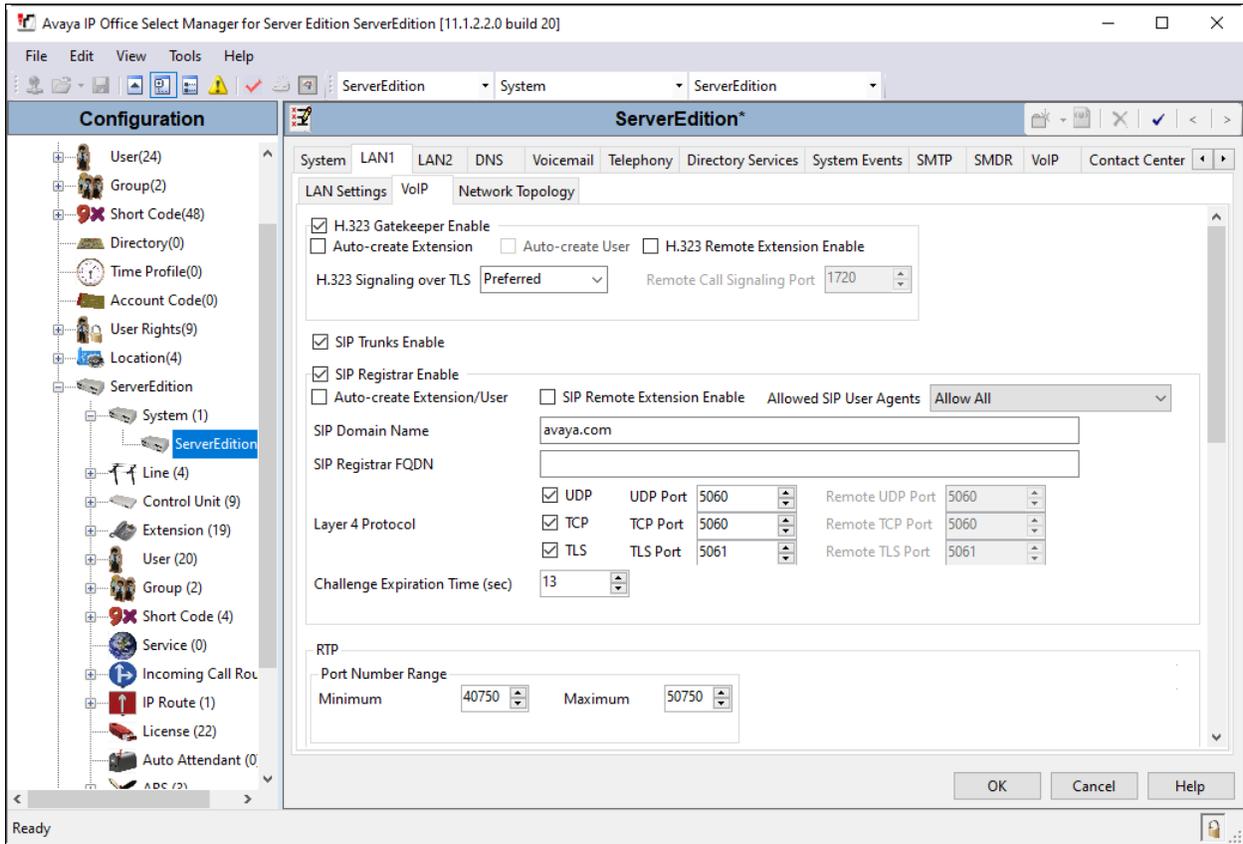
5.2. Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the System screen for the IP Office Server Edition in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the IP Address (e.g., 10.64.110.65), which will be used in **Section 6.3** to configure CTM-S2210-X/CTM-S2211-SPK.



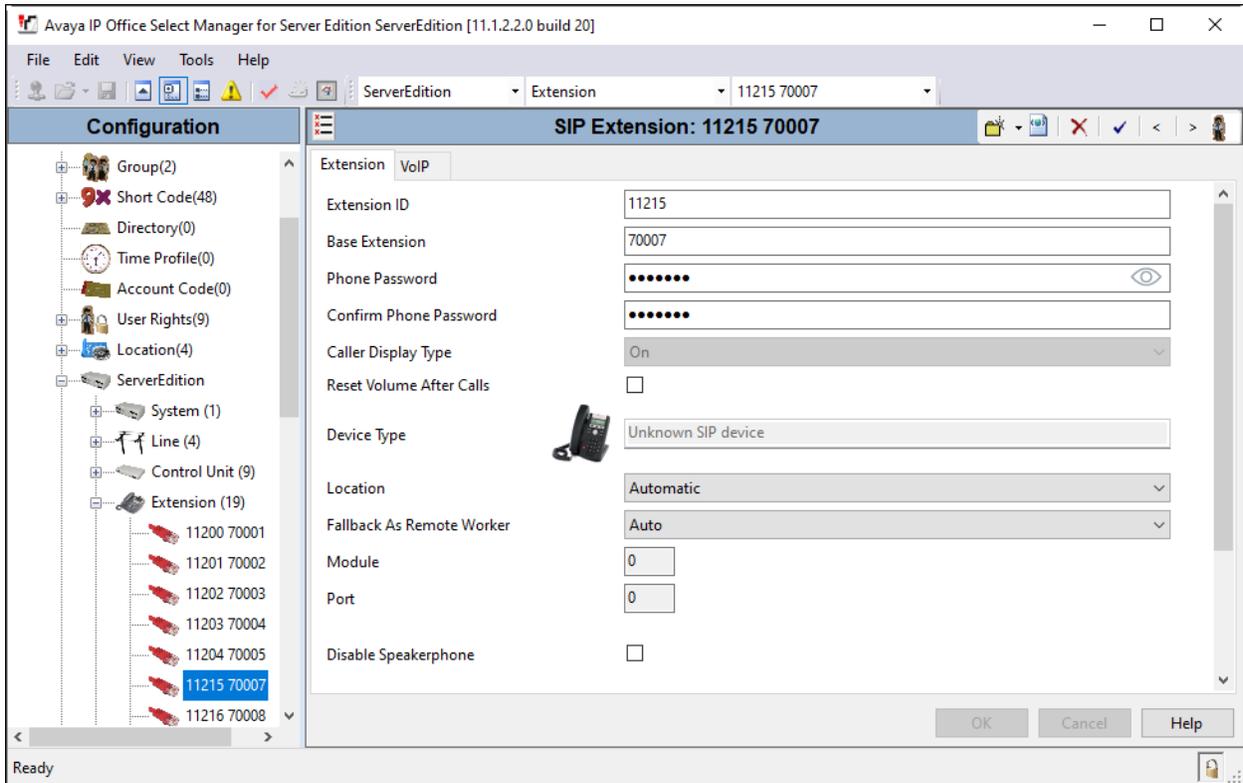
5.3. Administer SIP Registrar

Select the **VoIP** sub-tab. Ensure that **SIP Registrar Enable** is checked and enter a valid **Domain Name**. In the compliance testing, the **SIP Domain Name** field was set to *avaya.com*. TLS transport protocol was enabled for the **Layer 4 Protocol**, which was also used by CTM-S2210-X/CTM-S2211-SPK.

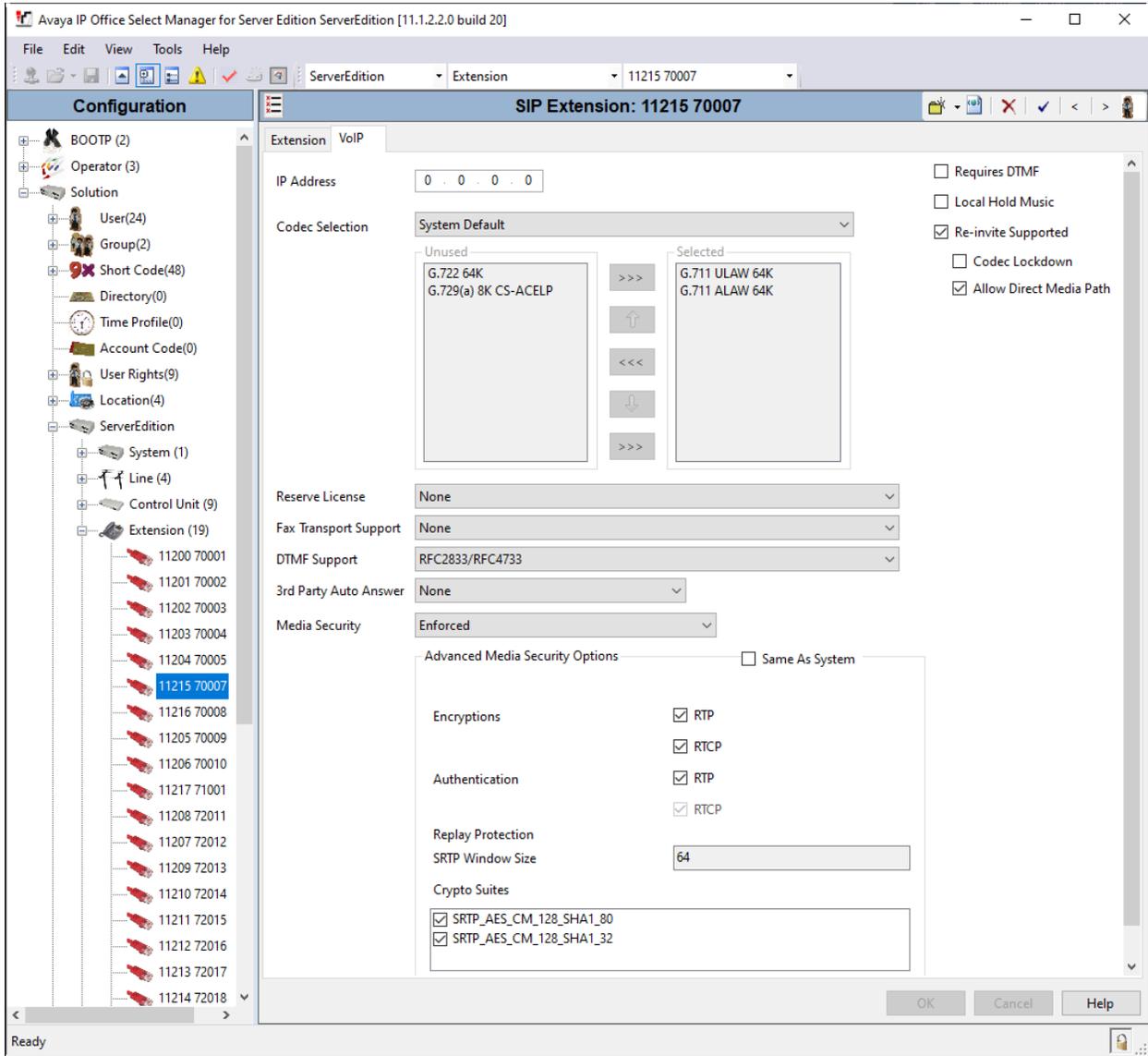


5.4. Administer SIP Extensions

From the configuration tree in the left pane, right-click on **Extension** and select **New** → **SIP Extension** from the pop-up list (not shown) to add a new SIP extension. Enter the desired extension for the **Base Extension** field as shown below. In this example, CTM-S2210-X was assigned extension *70007*. This is the extension that CTM-S2210-X will use to register with IP Office Server Edition. Enter an appropriate password. This will be used by CTM-S2210-X to register to IP Office Server.

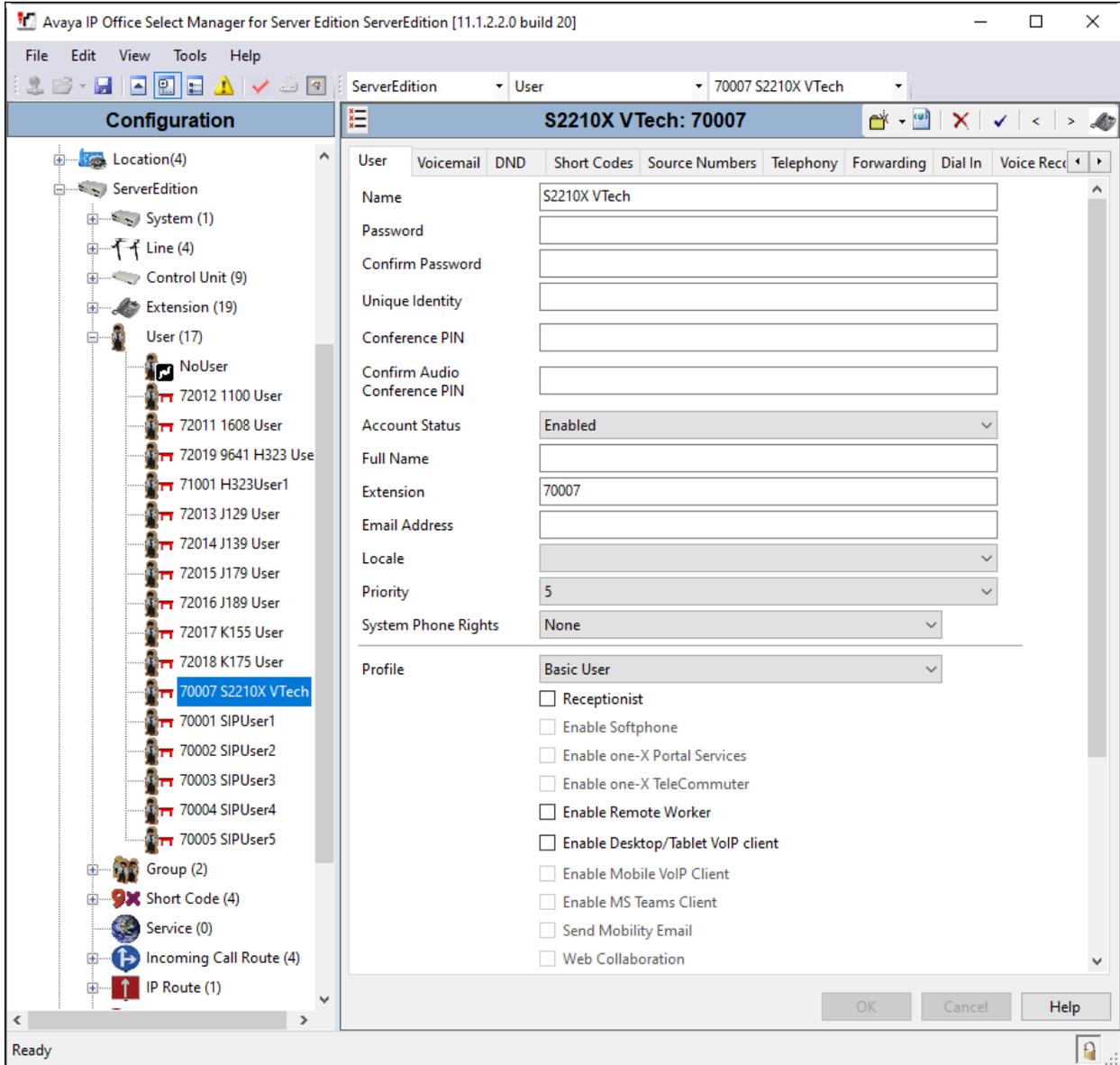


Select the **VoIP** tab. The codec selection shown below is configured with *G.711 ULAW* and *G.711 ALAW*. Enable **Allow Direct Media Path** so that audio/RTP may flow directly between two SIP endpoints without using media resources in Avaya IP Office Server Edition. Select *Enforced* for **Media Security** with **Advanced Media Security Options** as seen below.

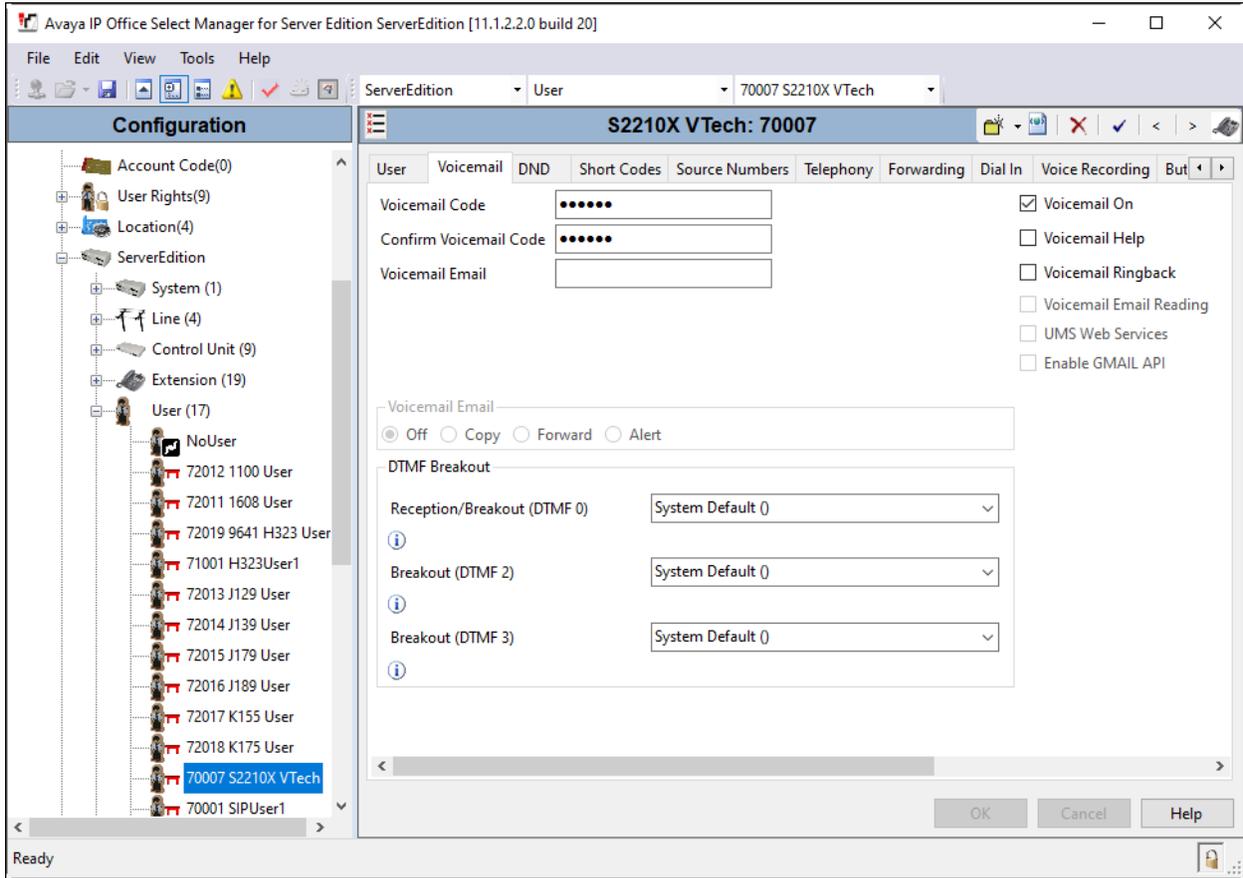


5.5. Administer SIP Users

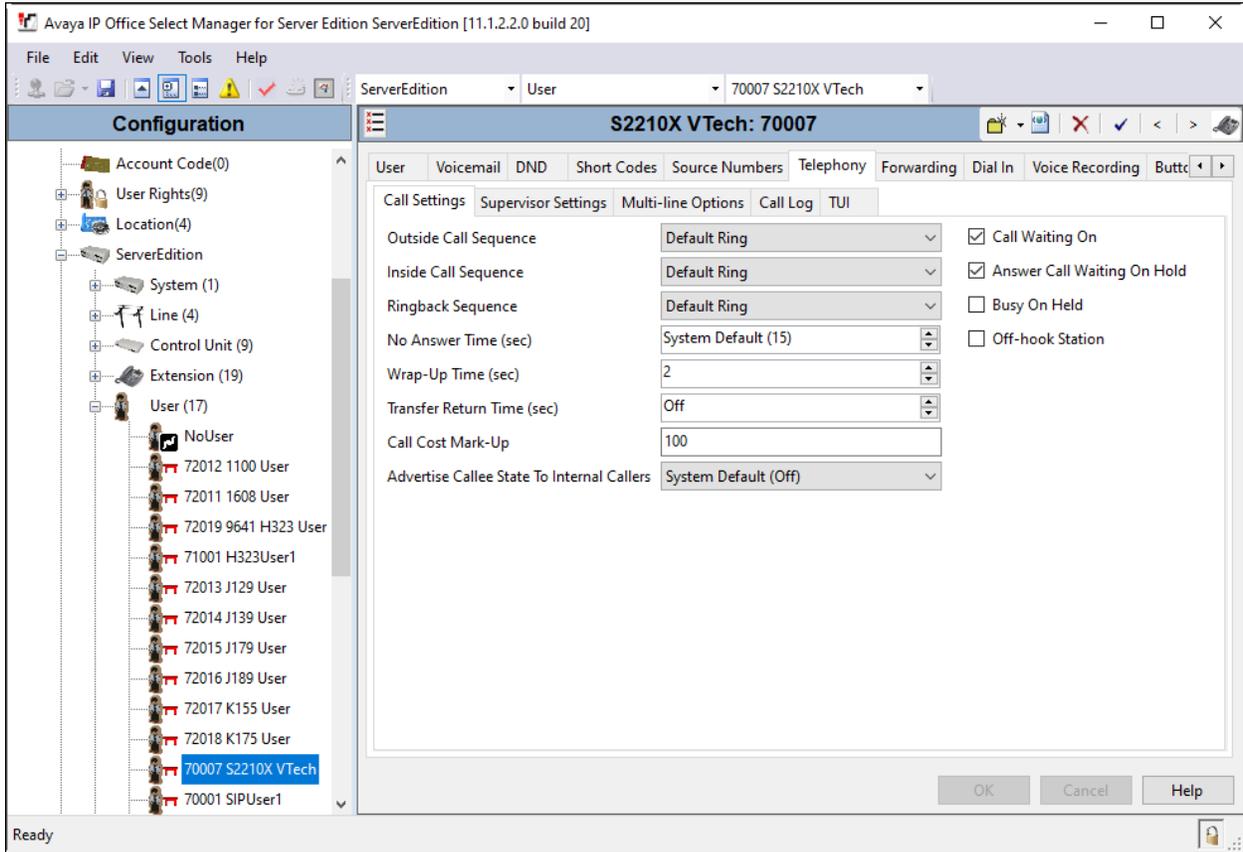
From the configuration tree in the left pane, right-click on **User** and select **New** from the pop-up list (not shown). Enter a value for the **Name** field (e.g., *S2210X VTech*). For the **Extension** field, enter the SIP extension from **Section 5.4** (e.g., *70007*).



Select the **Voicemail** tab and select **Voicemail On** to enable voicemail. Specify a **Voicemail Code** to be used when logging into voicemail.



Select the **Telephony** tab followed by the **Call Settings** sub-tab. Note the settings below for the user.



6. 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 IP Office Server Edition are as follows:

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

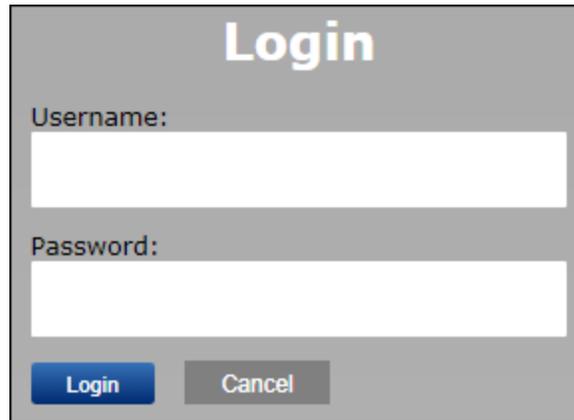
6.1. Configure IP Address

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 6.2**:

- 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>.

6.2. 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 6.1** using the URL **https://<IP address>**. The login prompt is displayed.



The image shows a web-based login form. At the top, the word "Login" is displayed in a large, bold, white font on a dark gray background. Below this, there are two input fields. The first is labeled "Username:" and the second is labeled "Password:". Both fields are empty and have a white background with a gray border. At the bottom of the form, there are two buttons: a blue button labeled "Login" and a gray button labeled "Cancel".

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



STATUS

System Status

STATUS

SYSTEM

NETWORK

SERVICING

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

6.3. Configure SIP Account

To register CTM-S2210-X/CTM-S2211-SPK to IP Office, 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:** Enter a descriptive string (e.g., *S2210X*).
- **Display Name:** Enter a desired display name (e.g., *S2210X IPO*).
- **User Identifier:** Enter An appropriate string (e.g., *70007*).
- **Authentication Name:** Enter the SIP extension from **Section 5.4** (e.g., *70007*).
- **Authentication Password:** Enter the password from **Section 5.4**.

The screenshot shows the vtech Hotel Phones configuration interface. The top left features the vtech logo and 'Hotel Phones' text. A navigation bar includes 'STATUS', 'SYSTEM', 'NETWORK', and 'SERVICING'. The left sidebar lists 'SYSTEM', '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 contains 'General Account Settings' with the following fields:

<input checked="" type="checkbox"/> Enable Account	
Account label:	<input type="text" value="S2210X"/>
Display Name:	<input type="text" value="S2210X IPO"/>
User Identifier:	<input type="text" value="70007"/>
Authentication Name:	<input type="text" value="70007"/>
Authentication Password:	<input type="password" value="....."/>
Dial Plan:	<input type="text" value="x+P"/>
Call Restriction Dial plan:	<input type="text"/>
Inter-Digit Timeout (secs):	<input type="text" value="3"/>
Line Type:	<input type="text" value="Private"/>
DTMF Method:	<input type="text" value="Auto"/>
Unregister After Reboot:	<input type="text" value="Disable"/>
Call Rejection Response Code	<input type="text" value="486"/>

Continuing on the same page, Under the **SIP Server** heading, enter the following:

- **Server Address:** IP Office Server Edition IP address (e.g., *10.64.110.65*).
- **Port:** *5061*

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

- **Server Address:** IP Office Server Edition IP address (e.g., *10.64.110.65*).
- **Port:** *5061*

	SIP Server	
	Server Address:	<input type="text" value="10.64.110.65"/>
	Port:	<input type="text" value="5061"/>
	Registration	
	Server Address:	<input type="text" value="10.64.110.65"/>
	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).

The screenshot displays a configuration interface with a blue sidebar on the left. The main content area is divided into several sections:

- Audio:** Contains seven dropdown menus for Codec Priority 1 through 7, with values G.711u, G.711a, G.729a/b, G.726, G.722, None, and iLBC respectively. It also includes checkboxes for 'Enable Voice Encryption (SRTP)' (checked) and 'Enable G.729 Annex B' (unchecked), a 'Preferred Packetization Time (ms)' dropdown set to 20, and a 'DTMF Payload Type' text field set to 101.
- Quality of Service:** Contains two text fields for DSCP (voice) and DSCP (signaling), both set to 46 and 26 respectively.
- Signaling Settings:** Contains a 'Local SIP Port' text field set to 5061 and a 'Transport' dropdown menu set to TLS.
- Voice:** Contains two text fields for 'Min Local RTP Port' and 'Max Local RTP Port', both set to 18000 and 19000 respectively.
- Voicemail Settings:** Contains a checked checkbox for 'Enable MWI Subscription'.

6.4. Install CA Certificate

Note: The CA certificate file must be installed in the VTech CTM-S2210-X/CTM-S2211-SPK Trusted Certificate store for validation of the IP Office 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.

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 <input type="checkbox"/>			
Total: 5	Issue to	Issue by	Expiration
<input type="checkbox"/>	Vtech Business Phone Intermediate CA	Vtech Business Phone Root CA	Feb 28 07:26:03 2036 GMT
<input type="checkbox"/>	thawte Primary Root CA - G3	thawte Primary Root CA - G3	Dec 1 23:59:59 2037 GMT
<input type="checkbox"/>	VeriSign Universal Root Certification Authority	VeriSign Universal Root Certification Authority	Dec 1 23:59:59 2037 GMT
<input type="checkbox"/>	DigiCert High Assurance EV Root CA	DigiCert High Assurance EV Root CA	Nov 10 00:00:00 2031 GMT
<input type="checkbox"/>	System Manager CA	System Manager CA	May 19 14:55:39 2047 GMT

Protected:

Delete Selected Entries Protect Selected Entries

Only accept trusted certificates

Save

Import Trusted Certificate:

6.5. Modify Codec Settings

Modify the codec settings by selecting **SYSTEM** (not shown) in the toolbar and **Account 1** (not shown) in the left hand side selections. Under the **Audio** heading, select the desired codecs ordered by priority. The default selections are 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"/>

Enable Voice Encryption (SRTP)

Enable G.729 Annex B

Preferred Packetization Time (ms):

DTMF Payload Type:

Click **Save**.

7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of IP Office and CTM-S2210-X/CTM-S2211-SPK.

7.1. Registration Status

Verify that CTM-S2210-X/CTM-S2211-SPK has successfully registered with IP Office. From a PC with **IP Office Admin Suite** installed, invoke **IP Office System Status**. Navigate to the CTM-S2210-X/CTM-S2211-SPK SIP extension and verify **Media Stream** is set to **SRTP**, **Layer 4 Protocol** is set to **TLS**, and **Current State** is shown as *Idle*.

The screenshot displays the AVAYA IP Office System Status application. The left sidebar contains a navigation menu with the following items: System, Alarms (4), Extensions (2) (with sub-items 70007 and 72013), Trunks (4), Active Calls, Resources, Voicemail, IP Networking, and Locations. The main content area is titled "Extension Status" and shows the following configuration details for extension 70007:

- Extension Number: 70007
- IP address: 192.168.4.11
- Standard Location: None
- Registrar: Primary
- Telephone Type: Unknown SIP Device
- User-Agent SIP header: Vtech Hotel SIP CTM-S2210-X 3.3.0.8-0
- Media Stream: SRTP
- Layer 4 Protocol: TLS
- Current User Extension Number: 70007
- Current User Name: S2210X VTech
- Forwarding: Off
- Twinning: Off
- Do Not Disturb: Off
- Message Waiting: Off
- Number of New Messages: 0
- Phone Manager Type: None
- SIP Device Features: REFER,UPDATE
- License Reserved: No
- Last Date and Time License Allocated: 12/8/2022 6:56:05 AM
- DTMF Required: No
- Packet Loss Fraction: [blank]
- Jitter: [blank]
- Round Trip Delay: [blank]
- Connection Type: [blank]
- Codec: [blank]
- Remote Media Address: [blank]

Below the configuration details is a table showing the current state of the extension:

Call Ref	Current State	Time in State	Calling Number or Called Number	Direction	Other Party on Call
	Idle	00:02:07			

At the bottom of the application, there are several buttons: Trace, Trace All, Pause, Ping, Call Details, Print..., and Save As... The status bar at the bottom right shows the time as 6:58:13 AM and the extension as Online.

Registration status can also be seen from the CTM-S2210-X/CTM-S2211-SPK web interface. Select **SYSTEM** from the toolbar, then **System Status** from the left-hand side list. Under **Account Status**, the account should show *Registered*.

The screenshot displays the vtech Hotel Phones web interface. At the top left is the vtech logo, and at the top center is the text "Hotel Phones" next to a globe icon. A navigation menu on the left includes "STATUS" and "System Status". A top navigation bar contains "STATUS", "SYSTEM", "NETWORK", and "SERVICING". The main content area is divided into two sections: "General" and "Account Status".

STATUS	SYSTEM	NETWORK	SERVICING
General			
Model:	CTM-S2210-X		
Serial Number:	7A700012167		
MAC Address:	A4:97:5C:96:64:F4		
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:	Registered		

7.2. Basic Calls

Establish a call between CTM-S2210-X/CTM-S2211-SPK and a local Avaya SIP desk phone. In **IP Office System Status**, navigate to the SIP extension and verify the **Current State** is *Connected* as shown below.

The screenshot displays the Avaya IP Office System Status web interface. The left sidebar contains a navigation menu with the following items: System, Alarms (4), Extensions (2) (with 70007 selected), Trunks (4), Active Calls, Resources, Voicemail, IP Networking, and Locations. The main content area is titled "IP Office System Status" and shows the "Extension Status" for extension 70007. The status is "Connected". Below the status information is a table showing call details.

Extension Number:	70007		
IP address:	192.168.4.11		
Standard Location:	None		
Registrar:	Primary		
Telephone Type:	Unknown SIP Device		
User-Agent SIP header:	Vtech Hotel SIP CTM-S2210-X 3.3.0.8-0		
Media Stream:	SRTP		
Layer 4 Protocol:	TLS		
Current User Extension Number:	70007		
Current User Name:	S2210X VTech		
Forwarding:	Off		
Twinning:	Off		
Do Not Disturb:	Off		
Message Waiting:	Off		
Number of New Messages:	0		
Phone Manager Type:	None		
SIP Device Features:	REFER,UPDATE		
License Reserved:	No		
Last Date and Time License Allocated:	12/8/2022 6:56:05 AM		
DTMF Required:	No		
Packet Loss Fraction:		Connection Type:	SRTP Direct Media
Jitter:		Codec:	G711 Mu
Round Trip Delay:		Remote Media Address:	192.168.4.6

Call Ref	Current State	Time in State	Calling Number or Called Number	Direction	Other Party on Call
497	Connected	00:00:27		Outgoing	Extn 72013, J129 User

At the bottom of the interface, there are buttons for Trace, Trace All, Pause, Ping, Call Details, Print..., and Save As... The status bar at the bottom right shows the time as 7:00:17 AM and the user as Online.

8. 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 IP Office Server Edition 11.1 and Avaya IP Office 500 V2 Expansion System 11.1. 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 IP Office Server Edition or Avaya IP Office 500 V2 Expansion System. 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**.

9. References

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

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

[1] *Administering Avaya IP Office using Manager*, Release 11.1, available at <http://support.avaya.com> as an HTML document.

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

[2] 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,

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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.
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