Avaya Solution & Interoperability Test Lab

Application Notes for Talkaphone VOIP-220 Series IP Call Stations with Avaya IP Office Server Edition - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Talkaphone VOIP-220 Series IP Call Stations 7.3.3.0 with Avaya IP Office Server Edition 11.1 and Avaya IP Office 500V2 Expansion System 11.1. Talkaphone VOIP-220 Series IP Call Stations are a family of indoor and outdoor-rated (ruggedized) VoIP emergency/information phones for use in locations such as parking facilities, college campuses, medical centers and industrial parks. Talkaphone VOIP-220 Series IP Call Stations register with Avaya IP Office as a SIP endpoint. For the compliance test, a Talkaphone VOIP-220C IP Call Station was used.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the 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 to integrate the Talkaphone VOIP-220 Series IP Call Stations 7.3.3.0 with Avaya IP Office Server Edition 11.1 and Avaya IP Office 500V2 Expansion System 11.1. Talkaphone VOIP-220 Series IP Call Stations are a family of indoor- and outdoor-rated (ruggedized) VoIP emergency/information phones for use in locations such as parking facilities, college campuses, medical centers and industrial parks. Talkaphone VOIP-220 Series IP Call Stations register with Avaya IP Office as a SIP endpoint. For the compliance test, a Talkaphone VOIP-220C IP Call Station was used. See **Attachment 1** for other models in the same series. Some models include a camera, but the video is not established as part of the voice call and was not tested.

Talkaphone VOIP-220 Series IP Call Stations incorporate Zenitel components and use the Zenitel GUI for configuration, under license from Zenitel USA, Inc.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Talkaphone VOIP-220 Series IP Call Stations, Avaya SIP / H.323 deskphones, and the PSTN, and exercising basic telephony features, such as hold, mute, transfer, and conference, from the Avaya IP deskphones. Additional telephony features, such as call forward and call coverage, initiated from Avaya IP deskphones were also verified.

The serviceability testing focused on verifying that the Talkaphone VOIP-220 Series IP Call Stations come back into service after re-connecting the Ethernet cable or rebooting the IP Call Station.

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 this DevConnect Application Note 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 this Application Note, the interface between Avaya systems and Talkaphone VOIP-220 Series IP Call Stations used TLS/SRTP encryption features.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of VOIP-220C with IP Office Server Edition or IP Office 500V2 Expansion.
- Calls between VOIP-220C and Avaya SIP / H.323 deskphones with Direct IP Media (Shuffling) enabled and disabled. Shuffling allows IP endpoints to send audio RTP packets directly to each other without using media resources in Avaya IP Office.
- Calls between VOIP-220C and the PSTN.
- Use of call button on VOIP-220C to place an outgoing call to Avaya IP deskphones and the PSTN.
- Use of Ring List on VOIP-220C to try multiple numbers for incoming calls.
- Playing a recording on VOIP-220C when a specific DTMF digit is entered by the connected party.
- G.711 and G.729 codec support.
- Support of TLS/SRTP using one-way authentication, TLS 1.2, and a secure PFS cipher.
- Support of UDP/RTP.
- Basic telephony features, including hold, mute, redial, call forwarding, transfer, and 3-way conference, initiated from Avaya IP deskphones.
- Call answer and termination on VOIP-220C via call button.
- Auto answer and manual answer on VOIP-220C.
- Call coverage on VOIP-220C.
- Long duration calls with VOIP-220C.
- Proper system recovery after a restart of VOIP-220C Station and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observation:

- Dialing short codes to activate telephony features are not applicable to Talkaphone IP Call Stations.

2.3. Support

For technical support and information on Talkaphone VOIP-220 Series IP Call Stations, contact Talkaphone Technical Support at:

- Phone: 1-773-539-1100
- Email: support@talkaphone.com
- Website: https://www.talkaphone.com/contact-support

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following products:

- Avaya IP Office Server Edition and Avaya IP Office 500 V2 Expansion connected via a SCN trunk and configured via Avaya IP Office Manager.
- PSTN connectivity provided by a SIP trunk on Avaya IP Office Server Edition and an ISDN-PRI trunk on Avaya IP Office 500 V2 Expansion System.
- Avaya 96x1 Series H.323 deskphones and Avaya J129 SIP Phones registered to Avaya IP Office Server Edition and Avaya IP Office 500 V2 Expansion.
- Talkaphone VOIP-220C IP Call Station registered to IP Office Server Edition or IP Office 500 V2 Expansion as a SIP endpoint.

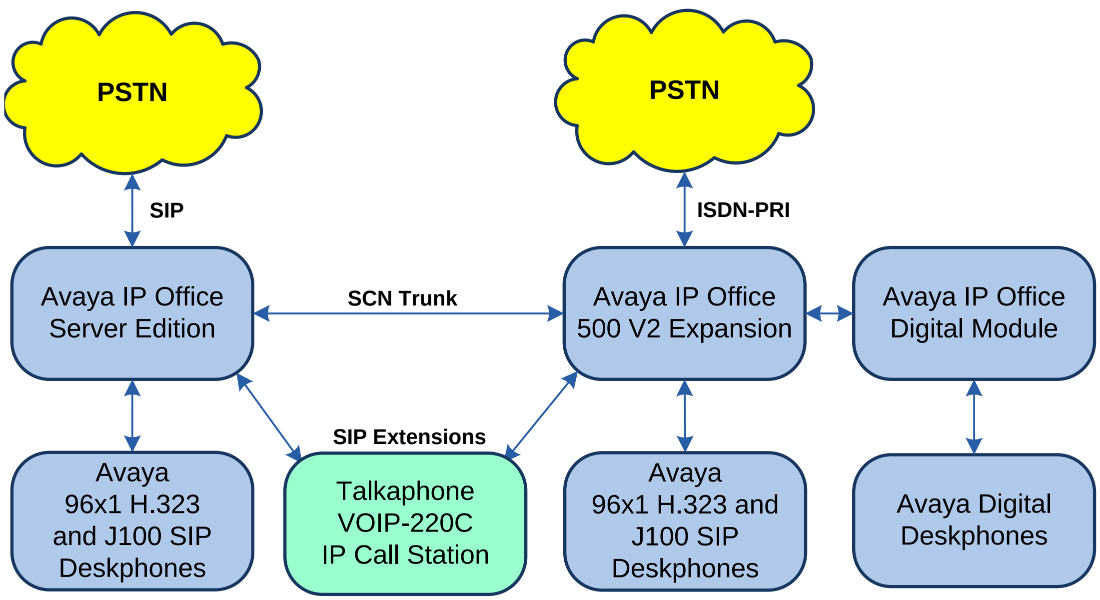


Figure 1: Avaya SIP Network with Talkaphone VOIP-220 Series IP Call Station

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office Server Edition	11.1.2.3.0 build 47
Avaya IP Office 500 V2 Expansion	11.1.2.3.0 build 47
Avaya 96x1 Series IP Deskphones	6.8.5.2.3 (H.323)
Avaya J100 Series IP Phones	4.0.10.3.2 (SIP)
Talkaphone VOIP-220C IP Call Station	7.3.3.0

Note: 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 procedure for configuring Avaya IP Office Server Edition. The procedure includes the following areas:

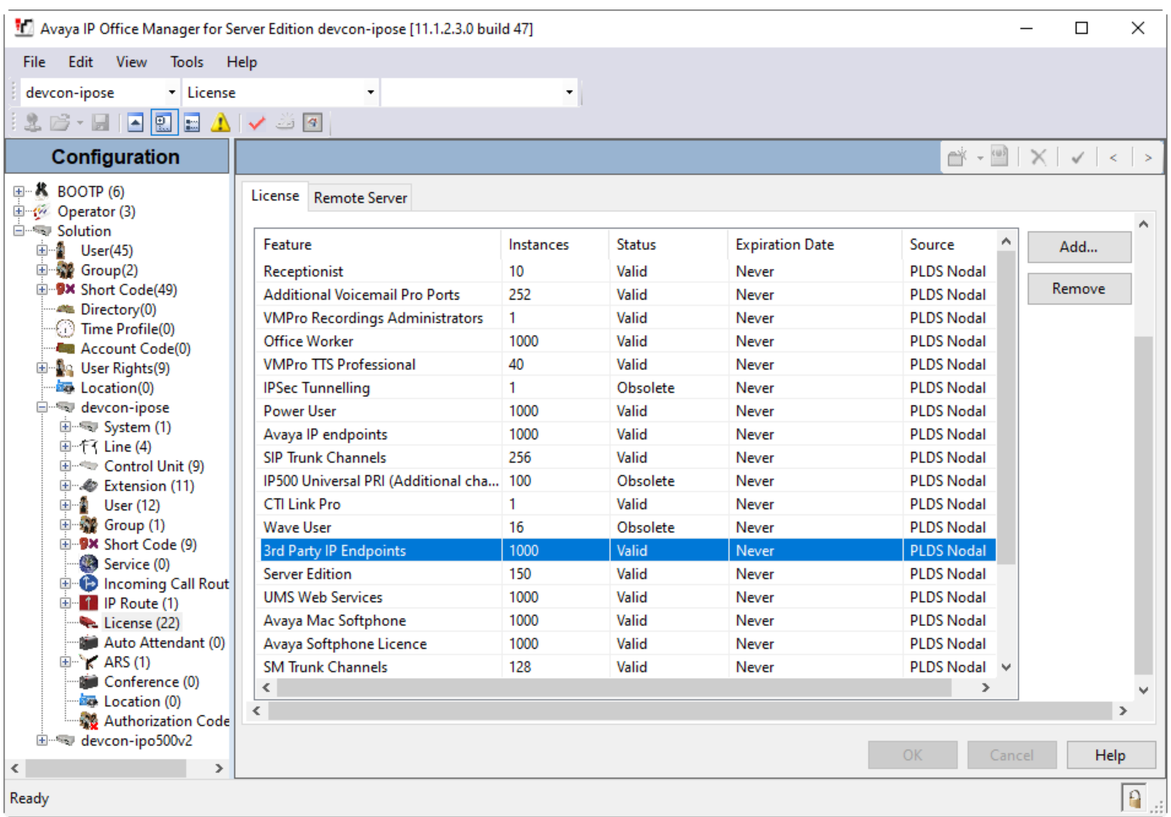
- Verify IP Office License
- Obtain LAN IP Address
- Administer SIP Registrar
- Administer SIP Extension
- Administer SIP User

Note: Integration of IP Office 500 V2 Expansion and call routing to the PSTN are outside the scope of these Application Notes.

5.1. Verify IP Office License

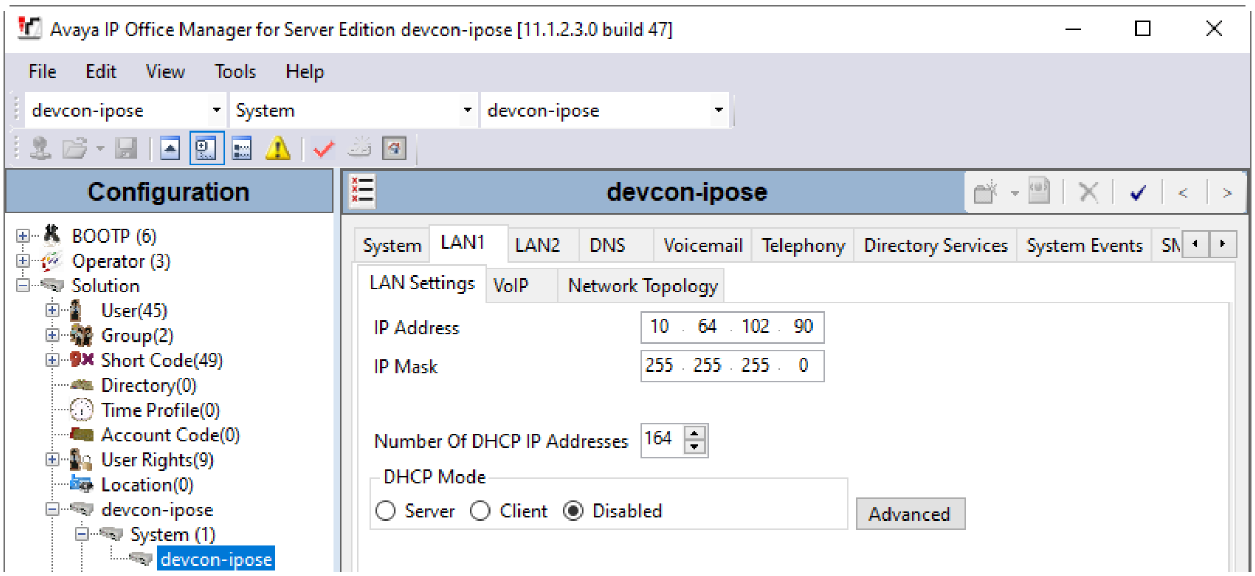
From a PC with Avaya IP Office Manager installed, select **Start → Programs → IP Office → Manager** to launch the Manager application. Select the required IP Office system and log in with the appropriate credentials.

The **Avaya IP Office Manager for Server Edition** screen is displayed. From the configuration tree in the left pane, select **License** to display the license screen in the right pane. Verify that the **License Status** is "Valid" for **3rd Party IP Endpoints**.



5.2. Obtain LAN IP Address

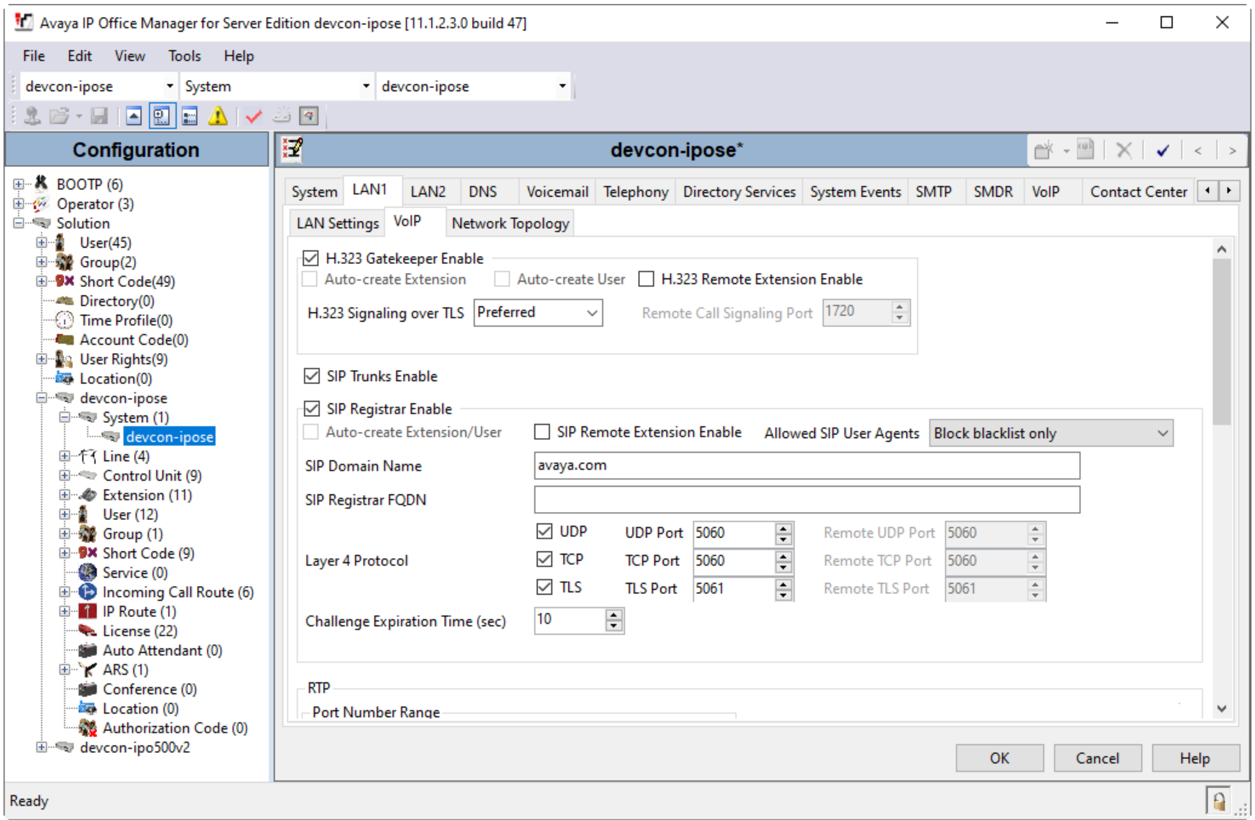
From the configuration tree in the left pane, select **System** to display the **System** screen for the IP Office Server Edition in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure VOIP-220C.



5.3. Administer SIP Registrar

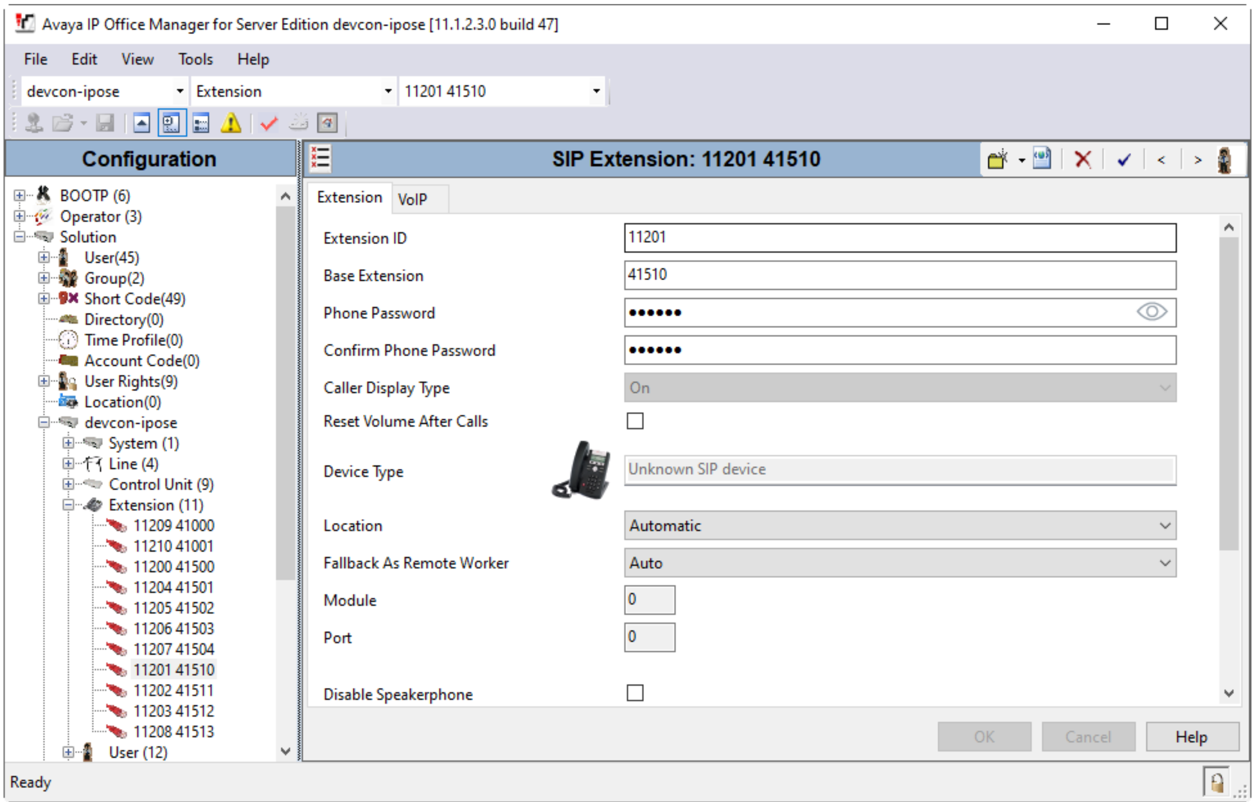
Select the **VoIP** sub-tab. Ensure that **SIP Registrar Enable** and that TLS transport is selected, which will be used by VOIP-220C, and enter a valid **SIP Domain Name** (e.g., *avaya.com*).

Note: VOIP-220C also support UDP transport. To use it instead of TLS, ensure that UDP transport is selected below.



5.4. Administer SIP Extension

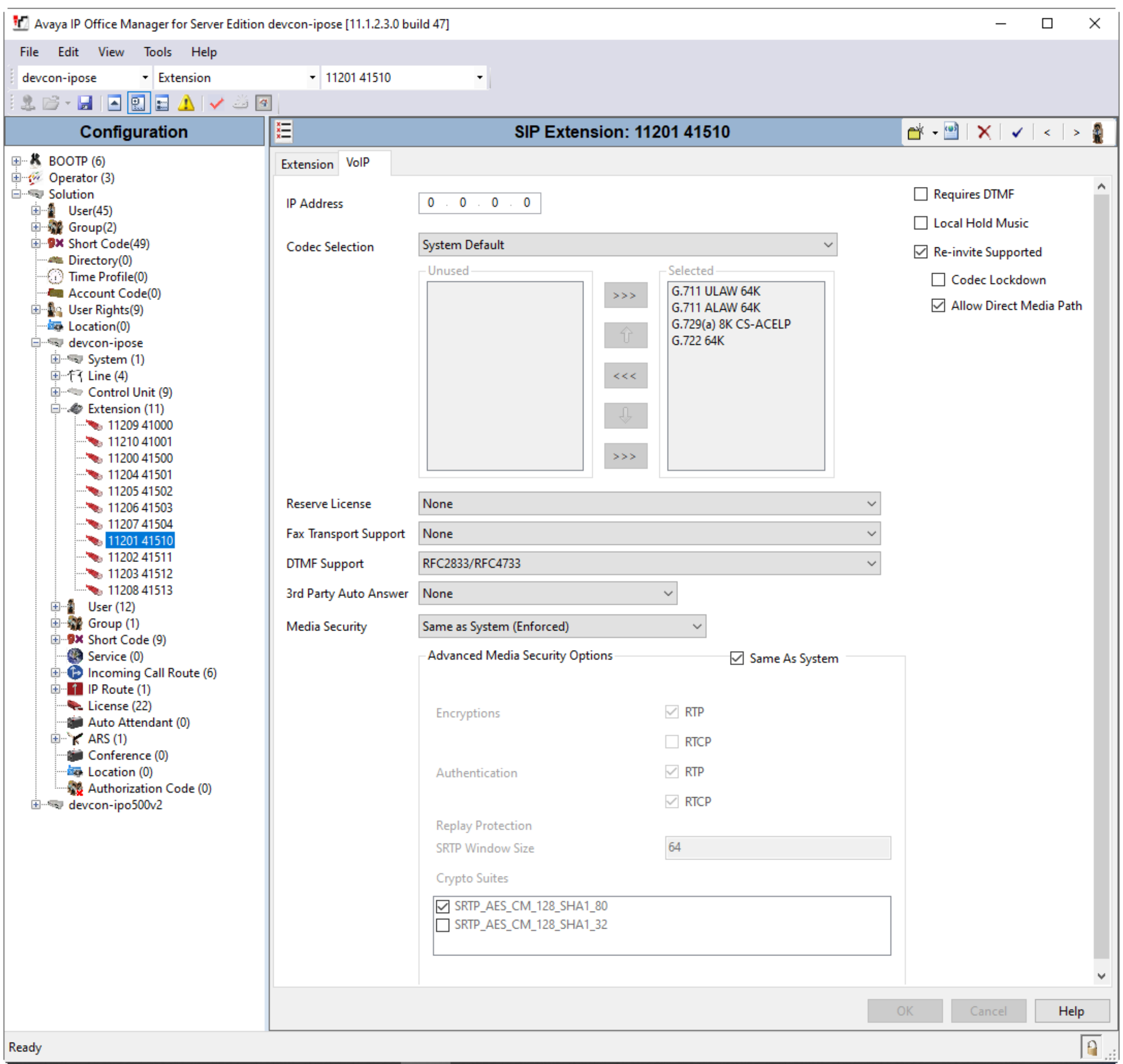
From the configuration tree in the left pane, right-click on **Extension** and select **New → SIP** from the pop-up list to add a new SIP extension (not shown). Enter the desired extension for the **Base Extension** field as shown below. In this example, VOIP-220C was assigned extension *41510*. Configure the **Phone Password** that will be used by VOIP-220C to register with IP Office Server Edition.



Select the **VoIP** tab. For **Codec Selection**, all the supported codecs may be selected. For the compliance test, G.711 and G.729 were verified with VOIP-220C. **Requires DTMF** may be set according to customer requirements. If VOIP-220C will require receiving DTMF, such as dialing a DTMF digit to play a recording, then **Requires DTMF** should be enabled. In this case, calls to Avaya H.323 phones will not use Direct IP Media. If VOIP-220C does not require receiving DTMF during a call, disable this option to allow shuffling to Avaya H.323 phones. Enable **Allow Direct Media Path** so that audio/RTP flows directly between two IP endpoints without using media resources in Avaya IP Office Server Edition.

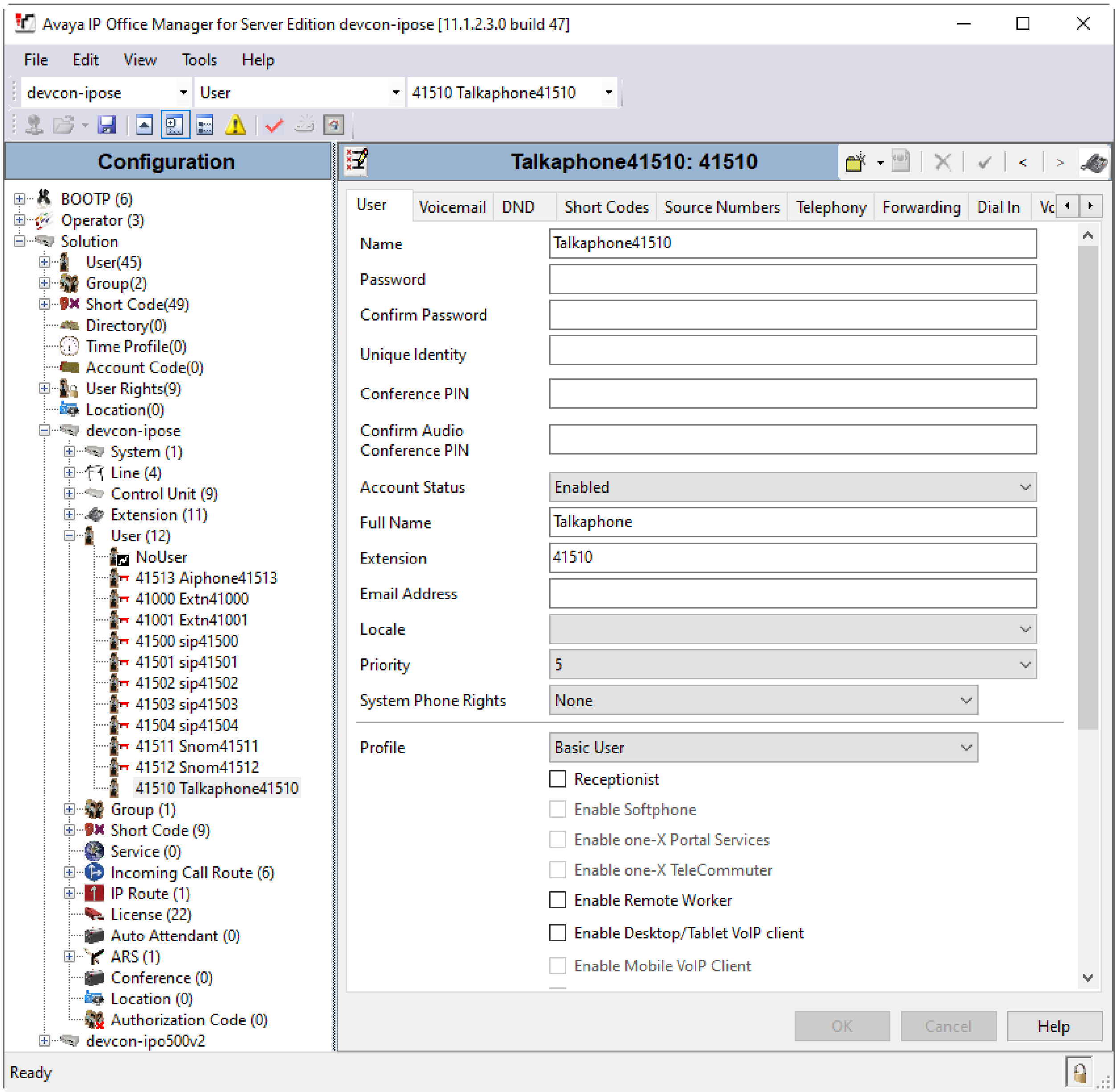
Media Security was enabled for VOIP-220C. The **Media Security** field should be set to *Enforced* to enforce SRTP for calls to VOIP-220C. Unencrypted RTP was used to match the configuration of VOIP-220C in **Section 6.3**.

Note: The Media Security section shown below was used for Avaya H.323 / SIP deskphones and the Web Socket SCN trunk between IP Office Server Edition and IP Office 500 V2 Expansion.

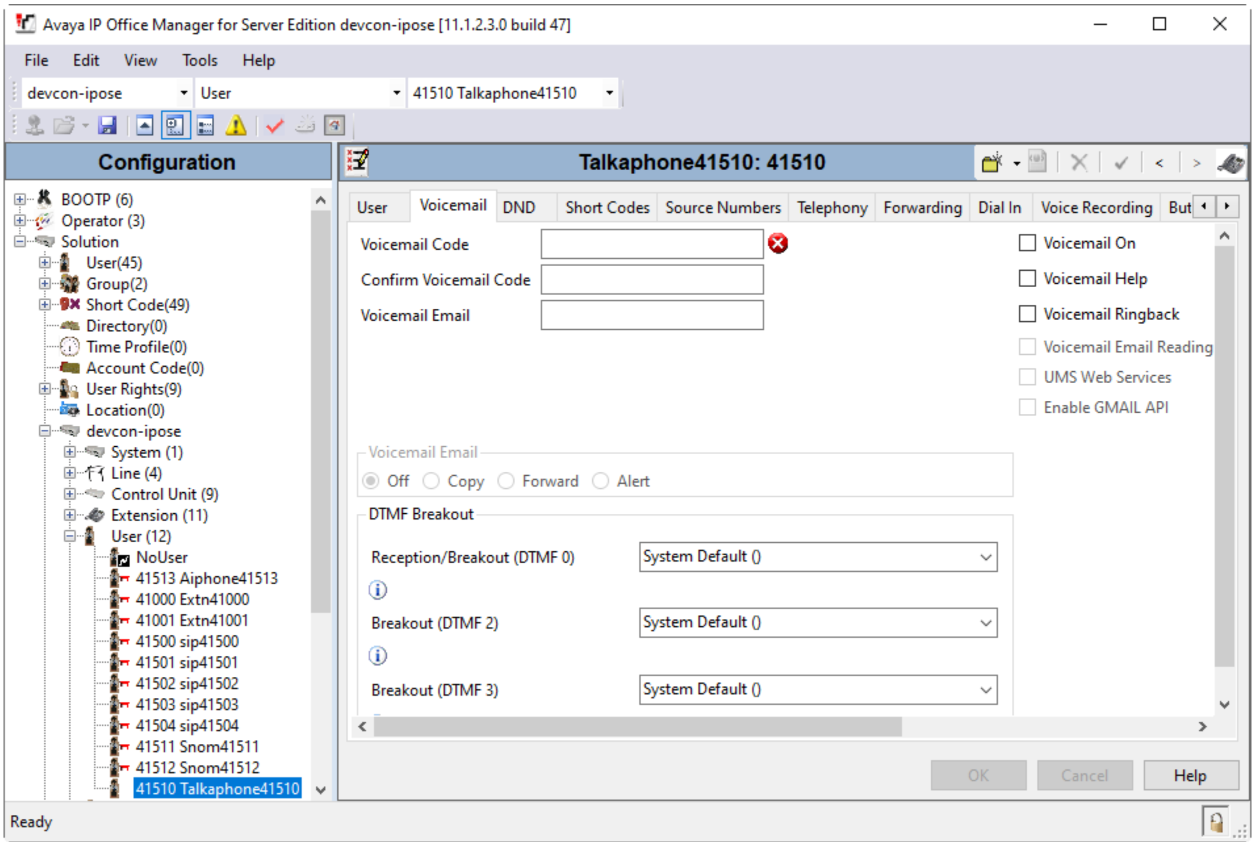


5.5. Administer SIP User

From the configuration tree in the left pane, right-click on **User** and select **New** from the pop-up list (not shown). Enter desired values for the **Name** and **Full Name** fields. For the **Extension** field, enter the SIP extension created in **Section 5.4**. The **Extension** field specifies the username that will be used by VOIP-220C to register with IP Office Server Edition.

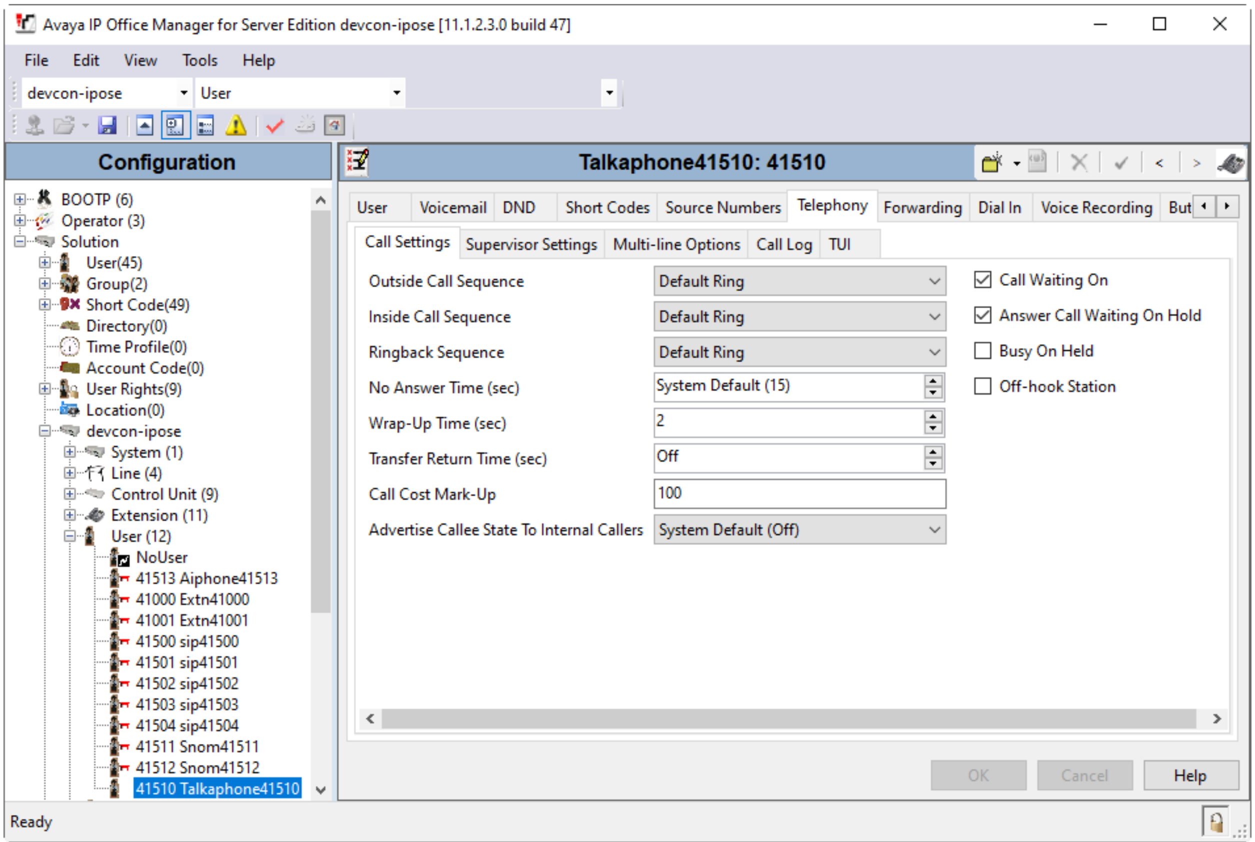


Select the **Voicemail** tab and disable voicemail for VOIP-220C.



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

Note: Call Waiting On is required to allow multiple calls to VOIP-220C; otherwise, subsequent incoming calls to VOIP-220C would be denied by IP Office Server Edition and return busy. Subsequent calls are queued and answered by VOIP-220C when it becomes available.



6. Configure Talkaphone VOIP-220 Series IP Call Station

This section covers the configuration of the Talkaphone VOIP-220 Series IP Call Station. The following procedures are covered:

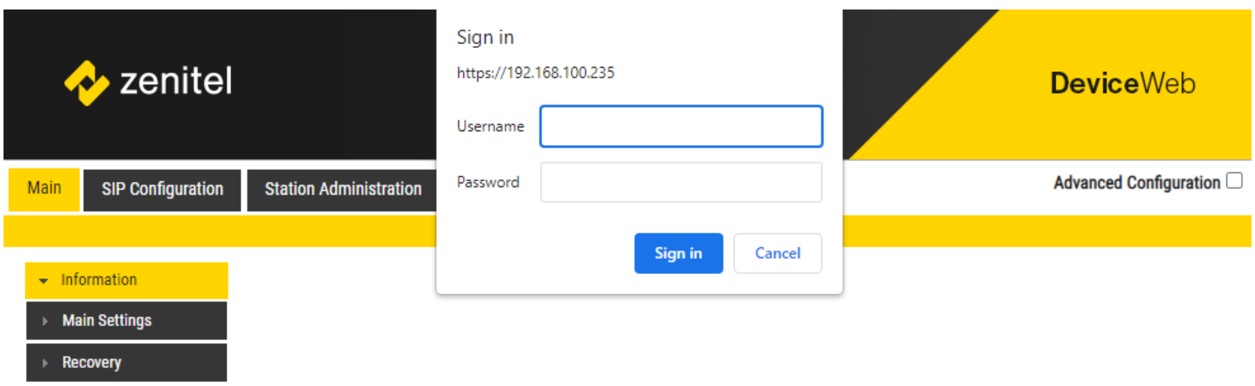
- Launching the Web Administration Interface
- Network Configuration
- SIP Configuration
- Configure Direct Access Keys
- Upload TLS Certificate

6.1. Launching the Web Administration Interface

Talkaphone IP Call Stations are pre-configured with the following default values:

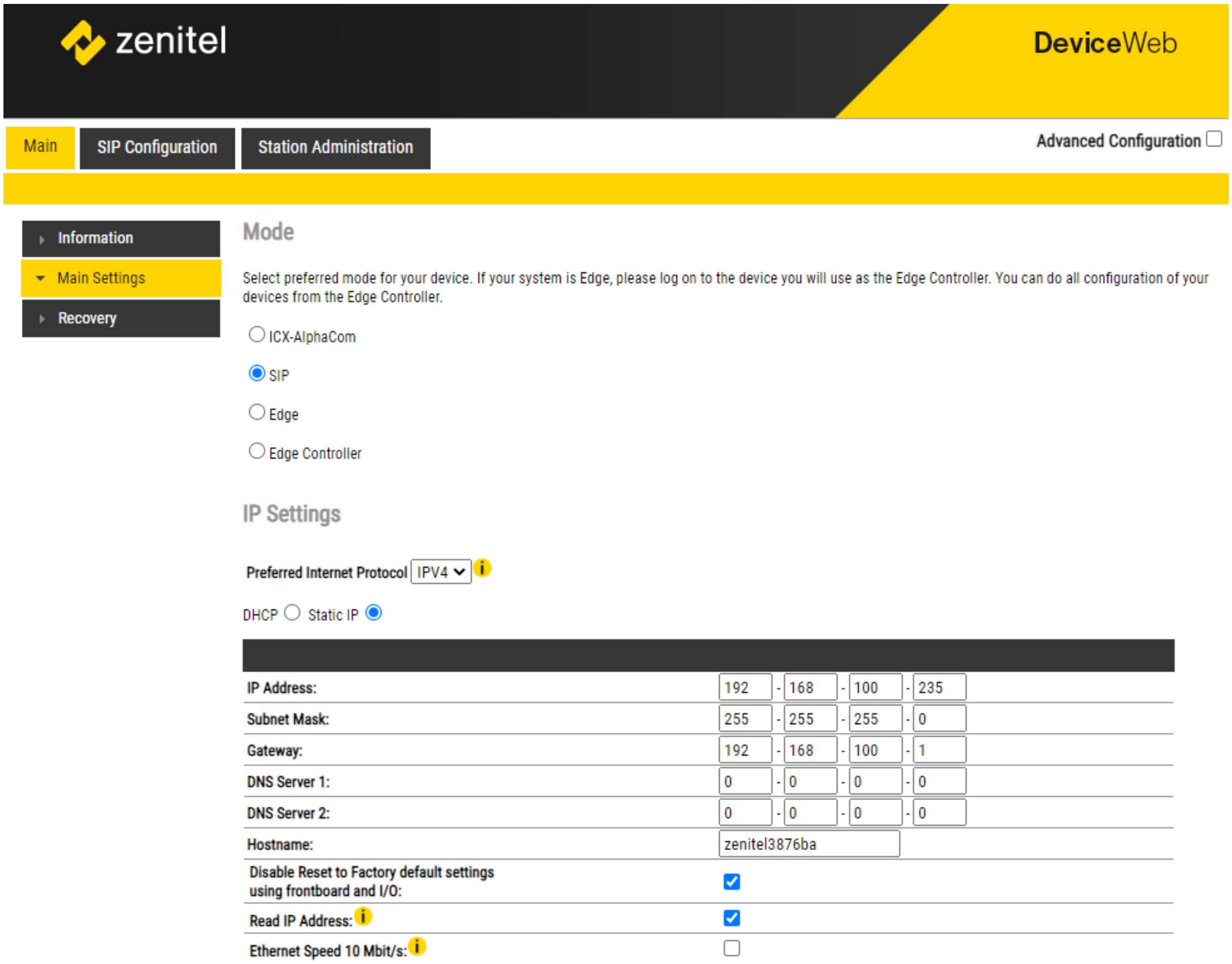
- **IP Address:** 192.168.1.10
- **Username:** admin
- **Password:** alphaadmin

Ensure that the administration PC and Talkaphone IP Call Station are connected to the LAN. Open a web browser and enter the default IP address of the Talkaphone IP Call Station in the URL field. The browser prompts for authentication. Log in with the appropriate credentials.



6.2. Network Configuration

To modify the IP network configuration of VOIP-220C navigate to the **Main → Main Settings** page. Verify that the **Mode** is set to *SIP*. Configure the IP settings so that it conforms to the customer network requirements.



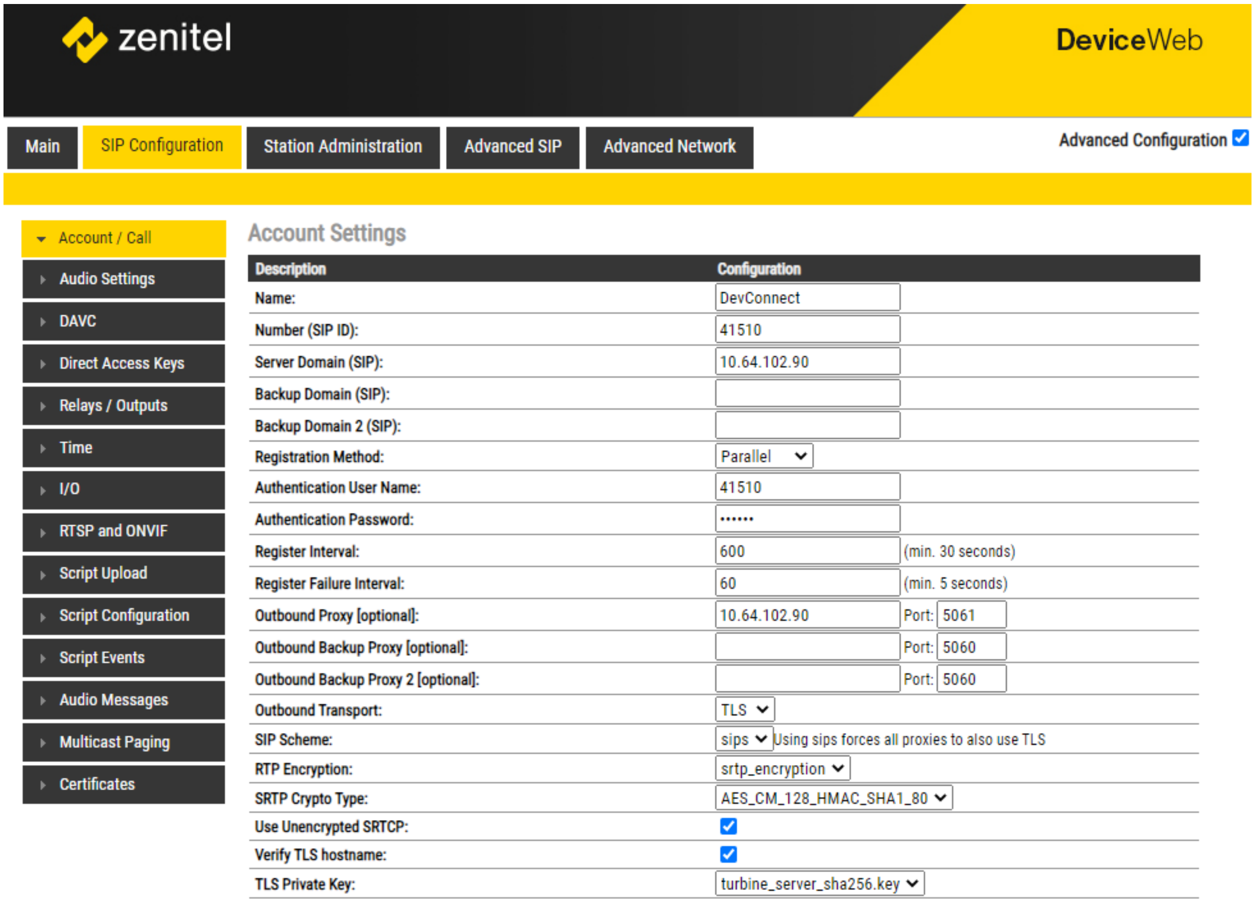
6.3. SIP Configuration

Navigate to **SIP Configuration → Account / Call** to configure the SIP settings of VOIP-220C. The following SIP configuration enables TLS/SRTP; however, UDP/RTP is also supported. Configure the following parameters.

Under **Account Settings**:

- **Name:** Specify a display name (e.g., *DevConnect*).
- **Number (SIP ID):** Specify the SIP number (e.g., *41510*) configured in **Section 5.4**.
- **Server Domain:** Specify the IP address of IP Office Server Edition (e.g., *10.64.102.90*).
- **Authentication User Name:** Specify the SIP number VOIP-220C (e.g., *41510*).
- **Authentication Password:** Specify the SIP password configured in **Section 5.4**.
- **Register Interval:** Set the SIP registration interval (e.g., *600* seconds).
- **Outbound Proxy 1 (optional):** Specify the IP address of IP Office Server Edition (e.g., *10.64.102.90*).
- **Port:** Specify the SIP port (e.g., *5061*).
- **Outbound Transport:** Set to *TLS*.
- **SIP Scheme:** Select *sip* or *sips*. *sips* was used for the compliance test.
- **RTP Encryption:** Select *srtp_encryption* to enable SRTP.
- **SRTP Crypto Type:** Select *AES_CM_128_HMAC_SHA1_80*. To match the **Crypto Suite** configured in IP Office in **Section 5.4**.
- **Use Unencrypted SRTCP:** Select *Unencrypted SRTCP*. Avaya H.323 phones do not support encrypted SRTCP.
- **Verify TLS hostname:** Enable TLS hostname verification.
- **TLS Private Key:** Accept default value of *turbine_server_sha256.key*.

Note: The TLS certificate is uploaded in **Section 6.5**.


zenitel
DeviceWeb
Main
SIP Configuration
Station Administration
Advanced SIP
Advanced Network
Advanced Configuration
Account / Call
Audio Settings
DAVC
Direct Access Keys
Relays / Outputs
Time
I/O
RTSP and ONVIF
Script Upload
Script Configuration
Script Events
Audio Messages
Multicast Paging
Certificates
Account Settings
Description
Configuration
Name:
DevConnect
Number (SIP ID):
41510
Server Domain (SIP):
10.64.102.90
Backup Domain (SIP):
Backup Domain 2 (SIP):
Registration Method:
Parallel
Authentication User Name:
41510
Authentication Password:
••••••
Register Interval:
600
(min. 30 seconds)
Register Failure Interval:
60
(min. 5 seconds)
Outbound Proxy [optional]:
10.64.102.90
Port: 5061
Outbound Backup Proxy [optional]:
Port: 5060
Outbound Backup Proxy 2 [optional]:
Port: 5060
Outbound Transport:
TLS
SIP Scheme:
sips
Using sips forces all proxies to also use TLS
RTP Encryption:
srtp_encryption
SRTP Crypto Type:
AES_CM_128_HMAC_SHA1_80
Use Unencrypted SRTCP:
Verify TLS hostname:
TLS Private Key:
turbine_server_sha256.key

In the **Call Settings** section, enable auto answer, if desired. To view additional call settings, select the **Advanced Configuration** checkbox as shown above.

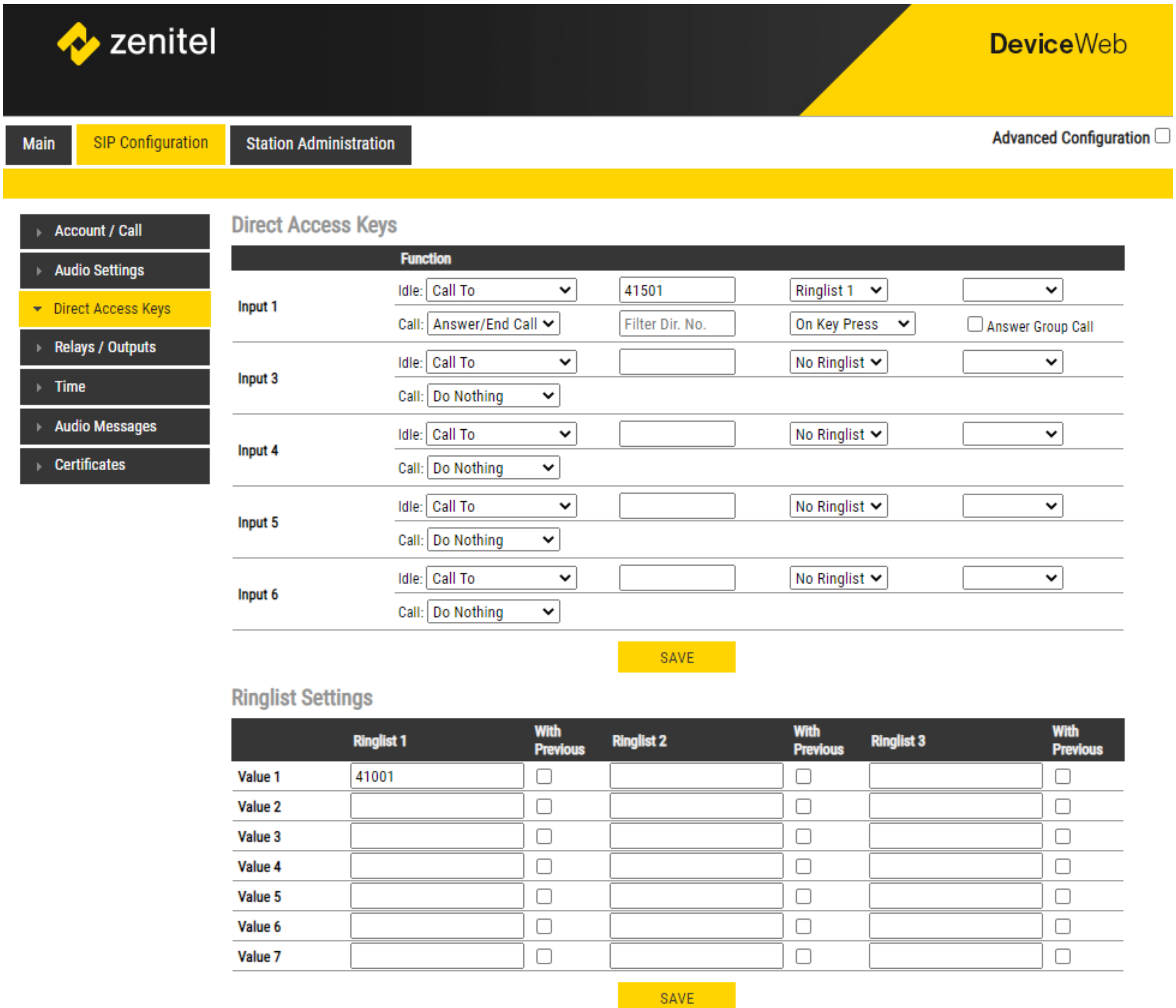
All of the default settings were used for the compliance testing, but this section also shows the codec configuration and is displayed for informational purposes.

Call Settings

Description	Configuration
Enable Auto Answer:	[x]
Auto Answer Delay:	0 seconds. Max 30 seconds.
Press and Hold Time:	0 seconds. Max 60 seconds. Defines how long a DAK key/Input must be pressed before the call is established.
Max Trying Time:	15 How long to wait on response before hanging up.
Max Ringing Time:	120 How long a call can be ringing before hanging up.
Max Conversation Time:	3600 How long a call can be in conversation before hanging up.
Max MP114 Speech Time:	0 How long between MP114 speech start/end before hanging up.
Max Queued Time:	20 How long a call can be queued before hanging up.
Max Queued Calls:	4 How many incoming calls can be queued. Max 5.
Use NAT Keep Alive:	[ ]
Dialing Method:	Enbloc Dialing
Enbloc Dialing Timeout:	No Timeout
DTMF method:	SIP INFO
Conversation Mode:	Duplex
PTT Mode:	Mic and speaker is controlled by PTT button
Resume Call Automatically:	[x] Resume Call On-Hold Automatically After Emergency Priority Ends
Remote Controlled Audio Direction:	[ ] (Received DTMF * to listen, DTMF # to talk, DTMF 0 for open duplex)
SIP Message Controlled Audio Direction:	[ ] (SIP MESSAGE controls audio direction)
Boost Volume on Push To Talk:	[ ]
Override Remote Push To Talk:	[ ]
Force Open Duplex Using DTMF:	-
Send DTMF */# with M key:	[x]
RTP Timeout value:	0 seconds. 0 = RTP Timeout Disabled.
SIP OPTIONS Timeout value:	0 seconds. 0 = SIP OPTIONS Timeout Disabled.
Codec g729:	Medium Priority
Codec g722:	High Priority
Codec g711a:	Medium Priority
Codec g711u:	Low Priority
Tone Volume:	0 (-1)=disabled, 0=default, [1..4]=[-22..-1]dB

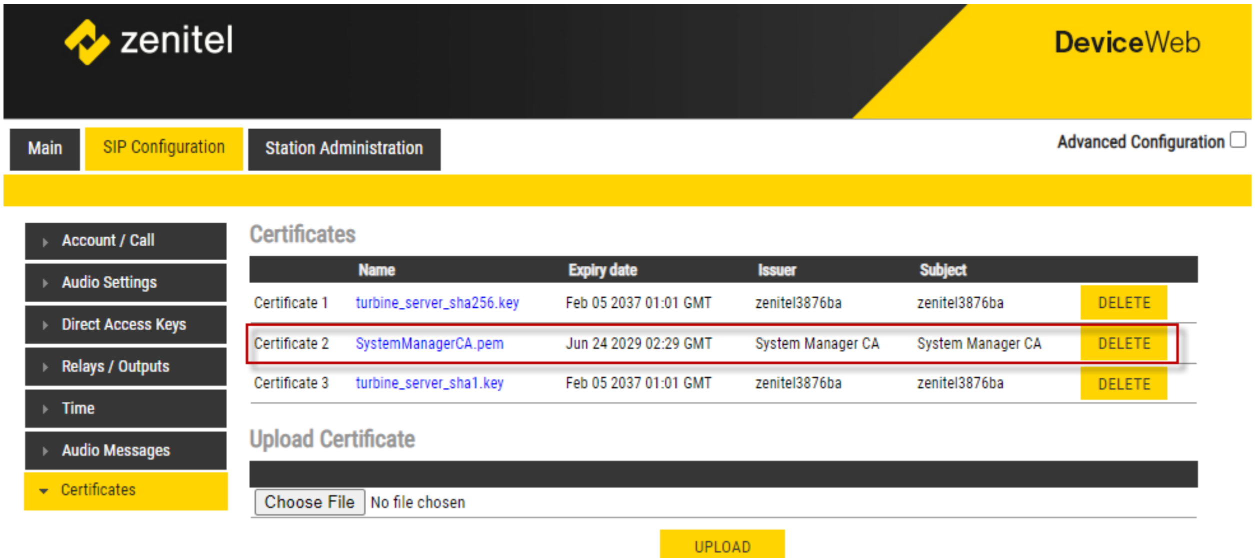
6.4. Configure Direct Access Keys

Navigate to **SIP Configuration → Direct Access Keys** to configure the behavior of VOIP-220C button. **Input 1** is configured to place a *Call To* the specified number, *41501*, when VOIP-220C is **Idle** and is associated with *Ringlist 1*. For incoming calls and active calls, the VOIP-220C button is configured to *Answer/End Call*. In the **Ringlist Settings** section, **Ringlist 1** is configured to try another number, *41001*, if the first call attempt to *41501* is not answered.



6.5. Upload TLS Certificate

To upload the TLS certificate to VOIP-220C, navigate to **SIP Configuration → Certificates** and upload the certificate in the **Upload Certificate** section. The installed certificate is shown below in the **Certificates** section. For the compliance test, the TLS certificate was obtained from System Manager CA.

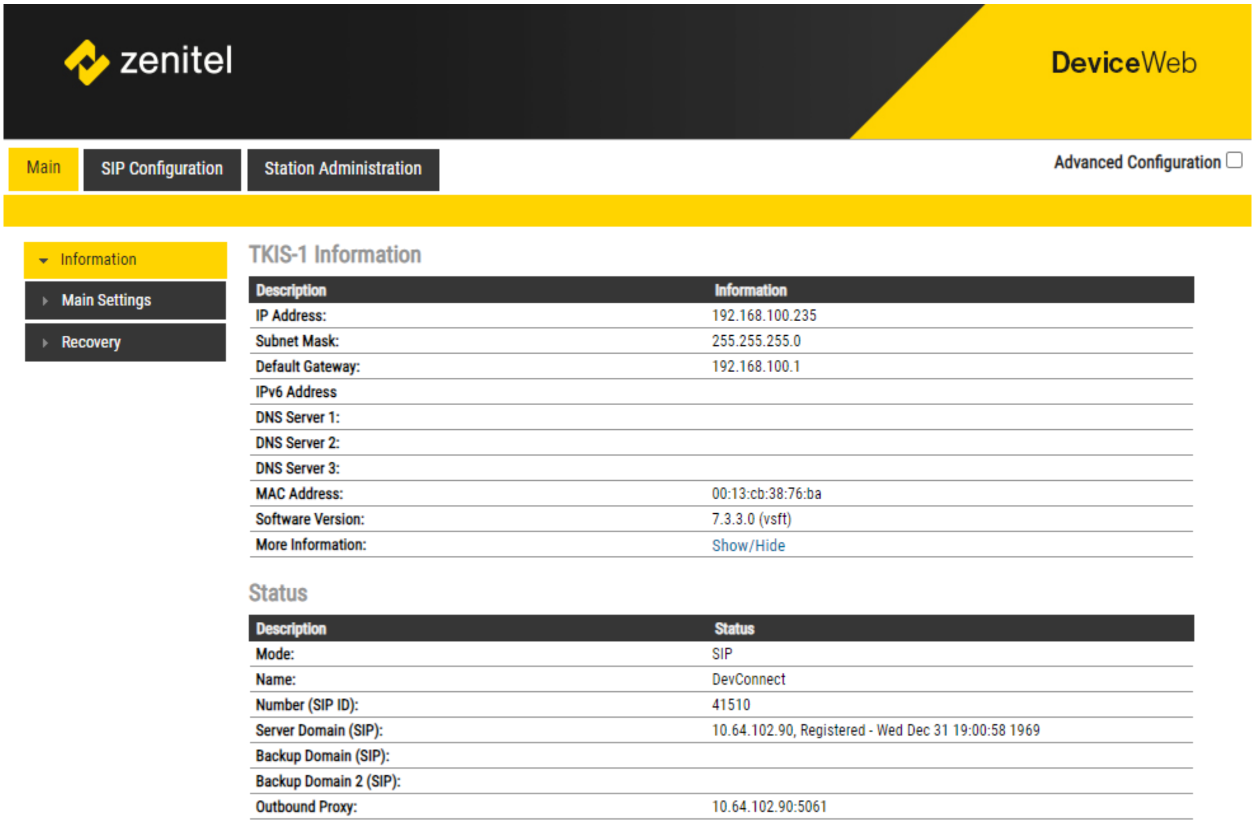


7. Verification Steps

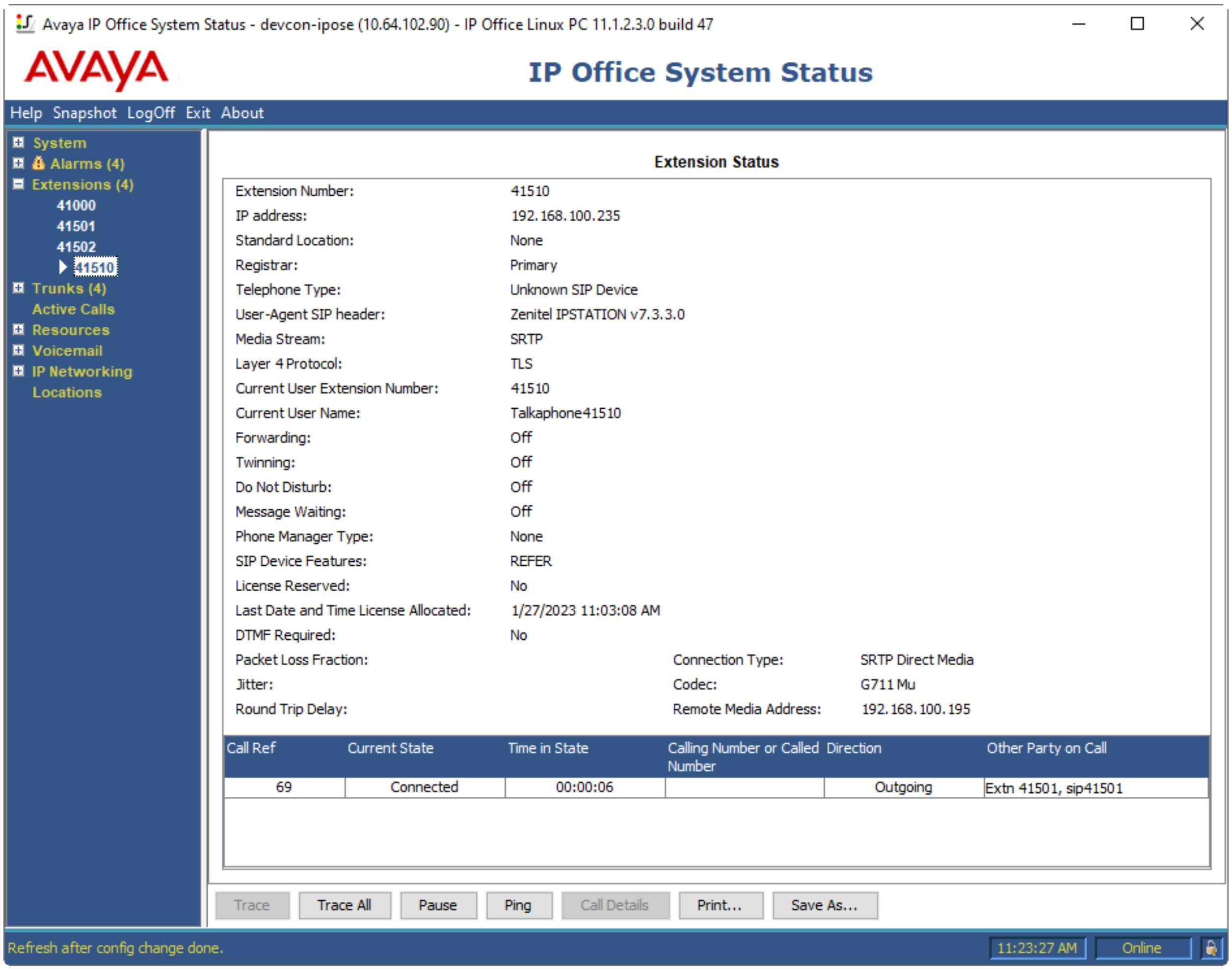
This section provides the tests that can be performed to verify proper configuration of the Talkaphone VOIP-220C IP Call Station with Avaya IP Office Server Edition.

1. Verify that VOIP-220C has successfully registered with IP Office Server Edition. Launch **IP Office System Status** and navigate to **Extensions → *<SIP Extension>*,** where *<SIP Extension>* is the VOIP-220C extension. Verify that the **Current State** is *Idle* and the **Layer 4 Protocol** is *TLS* as shown below.

2. Alternatively, the VOIP-220C registration status may be viewed on the VOIP-220C web interface. Navigate to **Main → Information** and verify that VOIP-220C is *Registered* in the **Server Domain (SIP)** field.



3. Place an incoming/outgoing call to to/from VOIP-220C and verify 2-way audio and proper call termination. The following **Extension Status** shows an active call on VOIP-220C using SRTP.



8. Conclusion

These Application Notes have described the administration steps required to integrate the Talkaphone VOIP-220 Series IP Call Stations with Avaya IP Office Server Edition. Talkaphone IP Call Stations successfully registered with IP Office Server Edition and basic telephony features were verified. All test cases passed with observations noted in **Section 2.2**.

9. Additional References

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

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

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.

TALKAPHONE

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F: 773-539-1241
talkaphone.com

February 20, 2023

Re: Common Platform for VOIP-200 Series Compact IP Call Stations

Avaya Inc.
350 Mt. Kemble Avenue
Morristown, NJ 07960

Attn: Avaya DevConnect Program

To Whom It May Concern:

Talkaphone's **VOIP-220 Series Compact IP Call Stations** incorporate a common SIP (Session Initiation Protocol) audio intercom PCBA (printed circuit board assembly) and firmware. This SIP audio intercom board is the Zenitel TKIS-1 VoIP Intercom Module and is incorporated under license from Zenitel USA, Inc.

Signage—The signage options outlined on p.7 of the VOIP-220 datasheet (revised on Aug. 17, 2022) only relates to the ADA-compliant features of the faceplate (i.e. the raised lettering and braille) and have no bearing with respect to any SIP interoperability testing.

It should be noted that the nomenclature for signage has carried over from the predecessor product, the VOIP-200 Series, and also applies to that product line.

Camera—Moreover, the camera included with certain VOIP-220 models also has no bearing with respect to the SIP interoperability testing. The camera is a standalone component and does not interact directly with the SIP audio intercom PCBA (i.e. the camera is packaged within a shared enclosure).

It should be noted that this camera arrangement has always been the case and also applies to the predecessor product, the VOIP-200 Series.

If there are further inquiries or concerns, please do not hesitate to contact us.

Sincerely,

Clarence Wong

Clarence Wong
Vice President – Product Management

Encl. (1)