



## **Application Notes for Talkphone VOIP-200 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 Talkphone VOIP-200 Series IP Call Stations with Avaya IP Office Server Edition. Talkphone VOIP-200 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. Talkphone VOIP-200 Series IP Call Stations support SIP (RFC 3261) and can operate as a paging/mass notification device via a standard SIP-based inbound call. Talkphone VOIP-200 Series IP Call Stations register with Avaya IP Office Server Edition as a SIP endpoint. For the compliance test, a Talkphone VOIP-201C3 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 Talkphone VOIP-200 Series IP Call Stations with Avaya IP Office Server Edition. Talkphone VOIP-200 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. Talkphone VOIP-200 Series IP Call Stations support SIP (RFC 3261) and can operate as a paging/mass notification device via a standard SIP-based inbound call. Talkphone VOIP-200 Series IP Call Stations register with Avaya IP Office Server Edition as a SIP endpoint. For the compliance test, a Talkphone VOIP-201C3 IP Call Station was used.

## 2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Talkphone VOIP-200 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, were also verified.

The serviceability testing focused on verifying that the Talkphone VOIP-200 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 Talkphone VOIP-200 Series IP Call Stations did not include use of any specific encryption features as requested by Talkphone.

## 2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of Talkphone IP Call Station with IP Office Server Edition.
- Inbound and outbound calls between Talkphone IP Call Station and Avaya SIP / H.323 Deskphones with Direct IP Media enabled and disabled.
- Inbound and outbound calls between the Talkphone IP Call Station and the PSTN.
- G.711 and G.729 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, transfer, and 3-way conference, initiated from the Avaya IP Deskphones.
- Use of recorded messages, speed-dial buttons, and number lists on the Talkphone IP Call Station.
- Proper system recovery after a restart of the Talkphone IP Call Station and loss of IP connectivity.

## 2.2. Test Results

All test cases passed with the following observation(s):

- Emergency calls cannot be terminated from Talkphone VOIP-200 Series IP Call Stations. This is by design. Emergency calls can only be disconnected by the far-end or upon expiration of the Call Conversation Timer. The destination of an emergency call shouldn't cover to voicemail. Talkphone VOIP-200 Series IP Call Stations dial a list of programmed numbers in a round-robin fashion. If the first number in the list does not answer, it will call the next number in line and will keep doing so until the destination answers the call or until the 'Call Conversation Timer' expires.
- Voice messages can only be recorded and played back when using G.711 codec. G.729 codec isn't supporting with recorded messages.
- When Talkphone IP Call Station establishes a call to an Avaya H.323 Deskphone, a call transfer (using the SIP REFER) from the Avaya H.323 Deskphone to an Avaya SIP Deskphone fails. Call transfers aren't expected to be common for this type of solution.
- Dialing short codes to activate telephony features are not applicable to Talkphone IP Call Stations.

## 2.3. Support

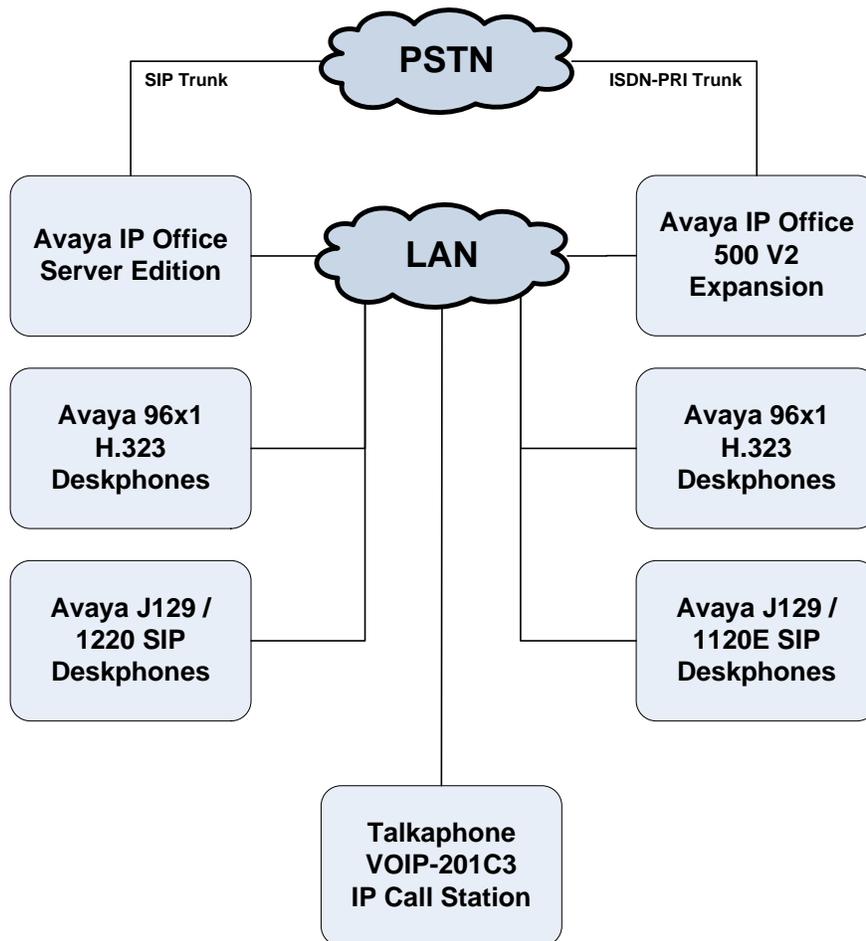
For technical support and information on Talkphone VOIP-200 Series IP Call Stations, contact Talkphone support at:

- Phone: 1-773-539-1100
- Email: [customerservice@talkphone.com](mailto:customerservice@talkphone.com)
- Website: <http://www.talkphone.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, Avaya J129 SIP Deskphones, and Avaya 1100/1200 Series SIP Deskphones registered to Avaya IP Office Server Edition and Avaya IP Office 500 V2 Expansion.
- Talkphone VOIP-200 Series IP Call Station registered to IP Office Server Edition as a SIP endpoint.



**Figure 1: Avaya SIP Network with Talkphone VOIP-200 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	11.0.4.0.0 build 74
Avaya IP Office 500 V2 Expansion	11.0.4.0.0 build 74
Avaya 96x1 Series IP Deskphones	6.8002 (H.323)
Avaya J129 SIP Deskphones	4.0.0.0.21 (21)
Avaya 1100/1200 SIP Deskphones	04.04.26.00
Talkaphone VOIP-200 Series IP Call Stations	2.4.1.40

### Notes:

- For the compliance test, a Talkaphone VOIP-201C3 IP Call Station was used.
- Compliance testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and when deployed with IP Office Server Edition in all configurations.

## 5. Configure Avaya IP Office Server Edition

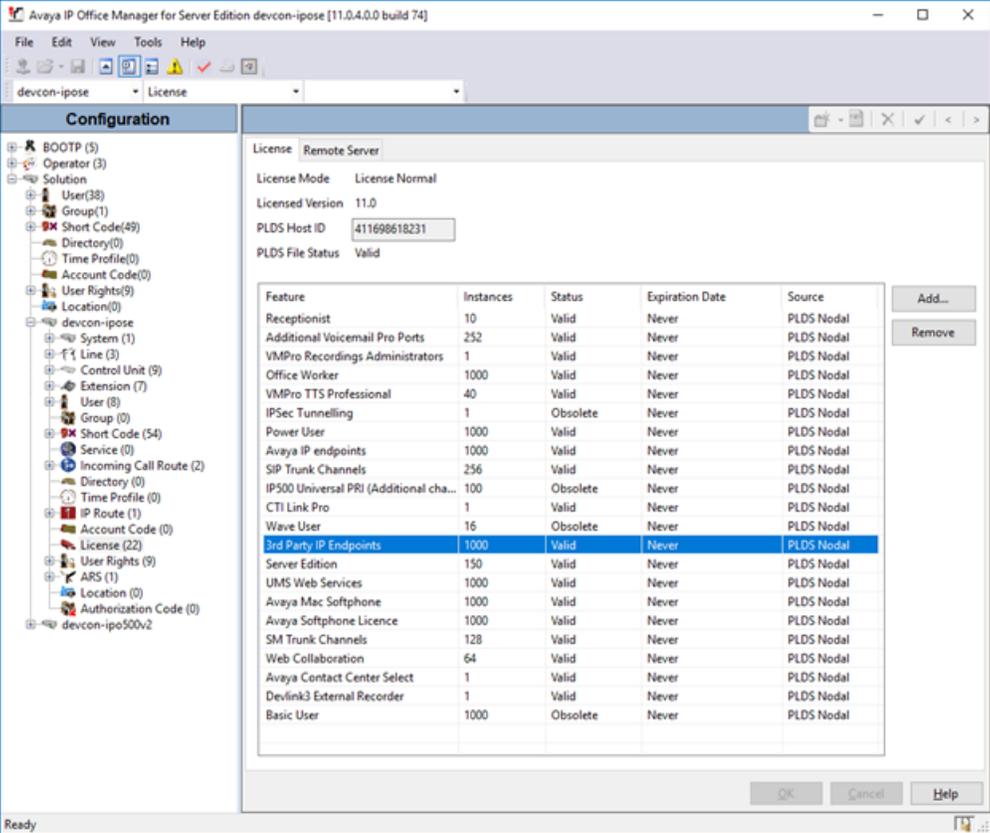
This section provides the procedure for configuring Avaya IP Office Server Edition. The procedure includes the following areas:

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

### 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 **3<sup>rd</sup> Party IP Endpoints**.

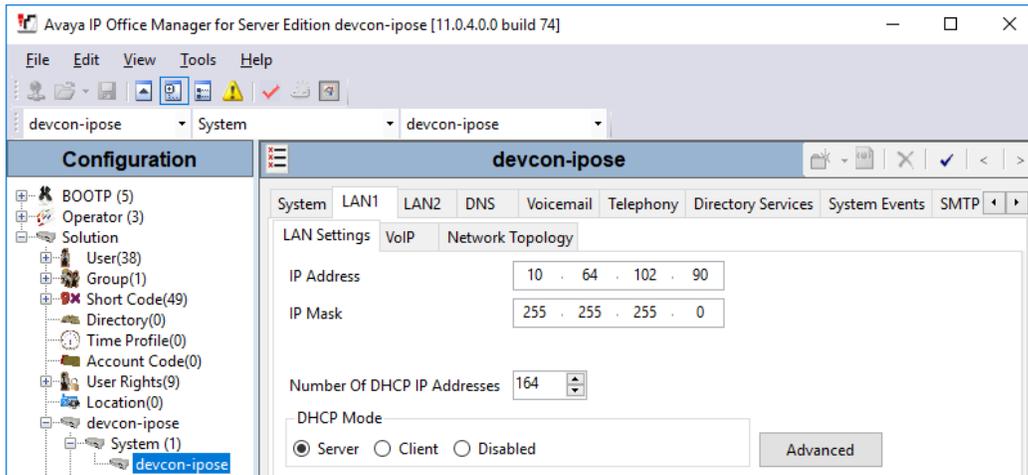


The screenshot shows the Avaya IP Office Manager for Server Edition interface. The left pane displays a configuration tree with 'License' selected. The right pane shows the 'License Remote Server' configuration. The 'License Mode' is 'License Normal', 'Licensed Version' is '11.0', 'PLDS Host ID' is '411698618231', and 'PLDS File Status' is 'Valid'. Below this, a table lists various features and their status.

Feature	Instances	Status	Expiration Date	Source
Receptionist	10	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	252	Valid	Never	PLDS Nodal
VMPro Recordings Administrators	1	Valid	Never	PLDS Nodal
Office Worker	1000	Valid	Never	PLDS Nodal
VMPro TTS Professional	40	Valid	Never	PLDS Nodal
IPSec Tunneling	1	Obsolete	Never	PLDS Nodal
Power User	1000	Valid	Never	PLDS Nodal
Avaya IP endpoints	1000	Valid	Never	PLDS Nodal
SIP Trunk Channels	256	Valid	Never	PLDS Nodal
IP500 Universal PRI (Additional cha...	100	Obsolete	Never	PLDS Nodal
CTI Link Pro	1	Valid	Never	PLDS Nodal
Wave User	16	Obsolete	Never	PLDS Nodal
<b>3rd Party IP Endpoints</b>	<b>1000</b>	<b>Valid</b>	<b>Never</b>	<b>PLDS Nodal</b>
Server Edition	150	Valid	Never	PLDS Nodal
UMS Web Services	1000	Valid	Never	PLDS Nodal
Avaya Mac Softphone	1000	Valid	Never	PLDS Nodal
Avaya Softphone Licence	1000	Valid	Never	PLDS Nodal
SM Trunk Channels	128	Valid	Never	PLDS Nodal
Web Collaboration	64	Valid	Never	PLDS Nodal
Avaya Contact Center Select	1	Valid	Never	PLDS Nodal
Devlink3 External Recorder	1	Valid	Never	PLDS Nodal
Basic User	1000	Obsolete	Never	PLDS Nodal

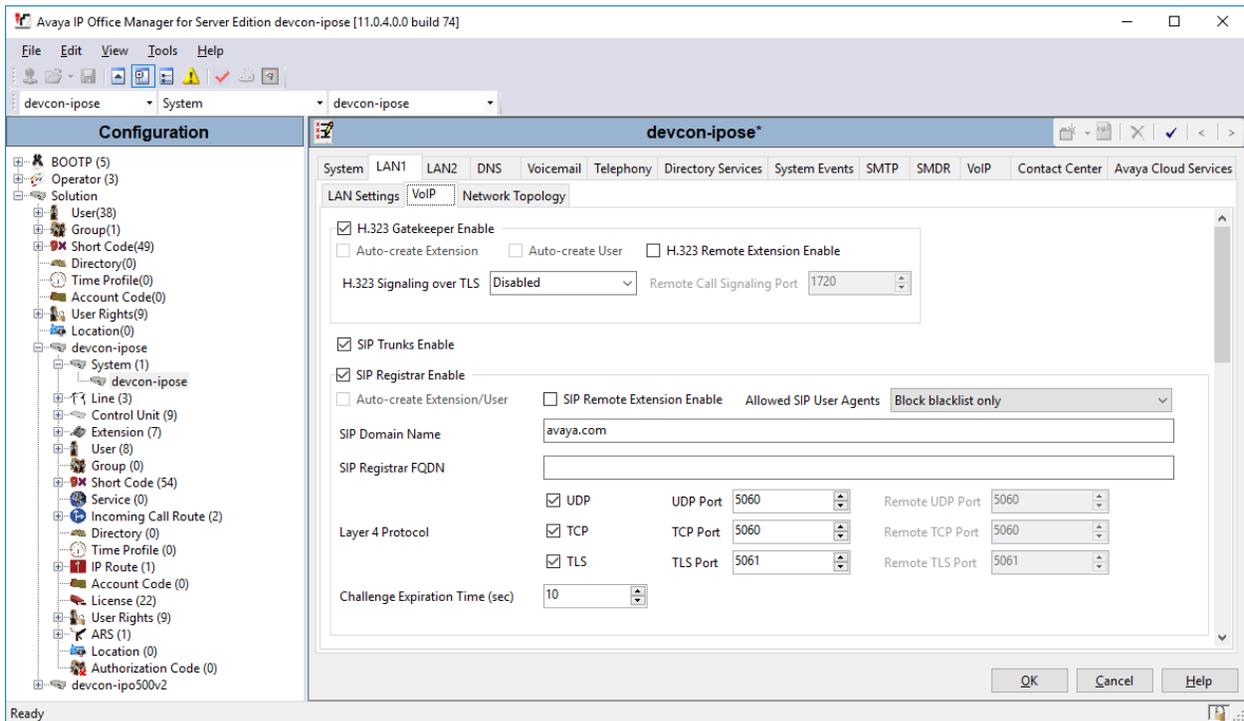
## 5.2. Obtain LAN IP Address

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 Talkphone IP Call Station.



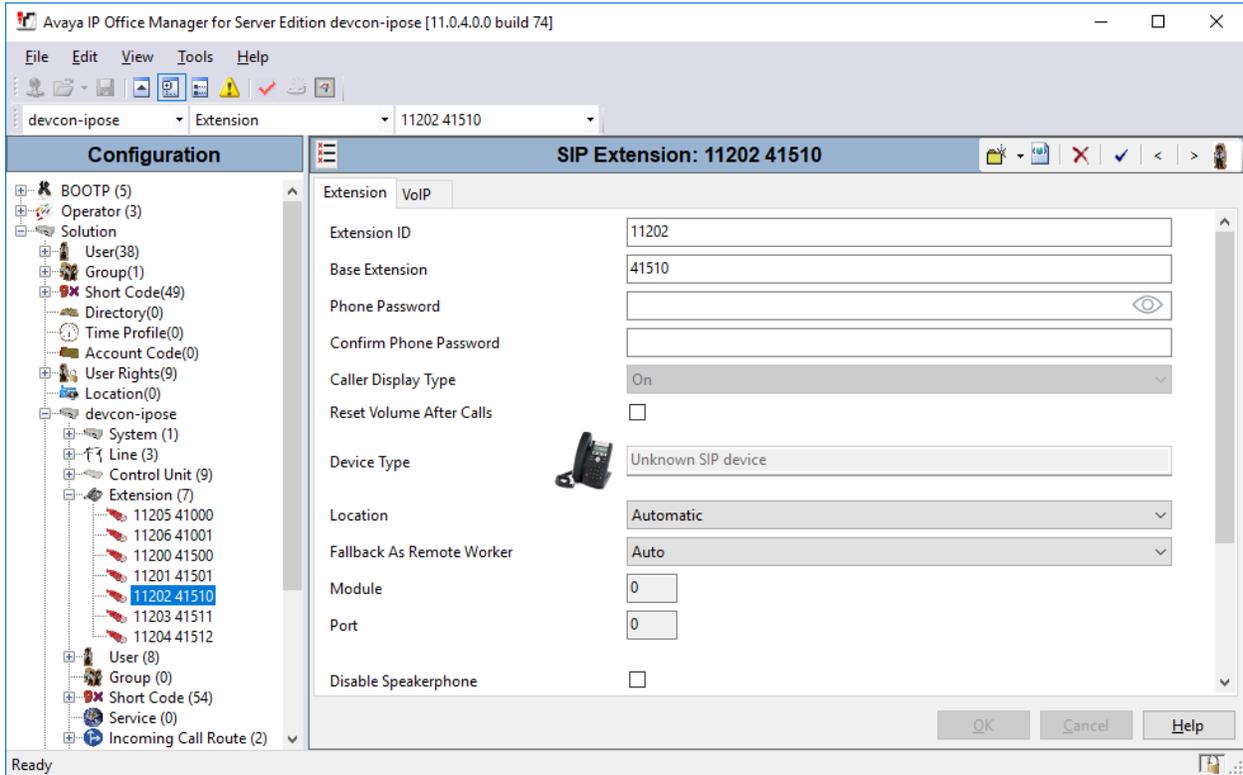
## 5.3. Administer SIP Registrar

Select the **VoIP** sub-tab. Ensure that **SIP Registrar Enable** and that UDP transport is selected, which will be used by Talkphone IP Call Station. Also, enter a valid **Domain Name**.

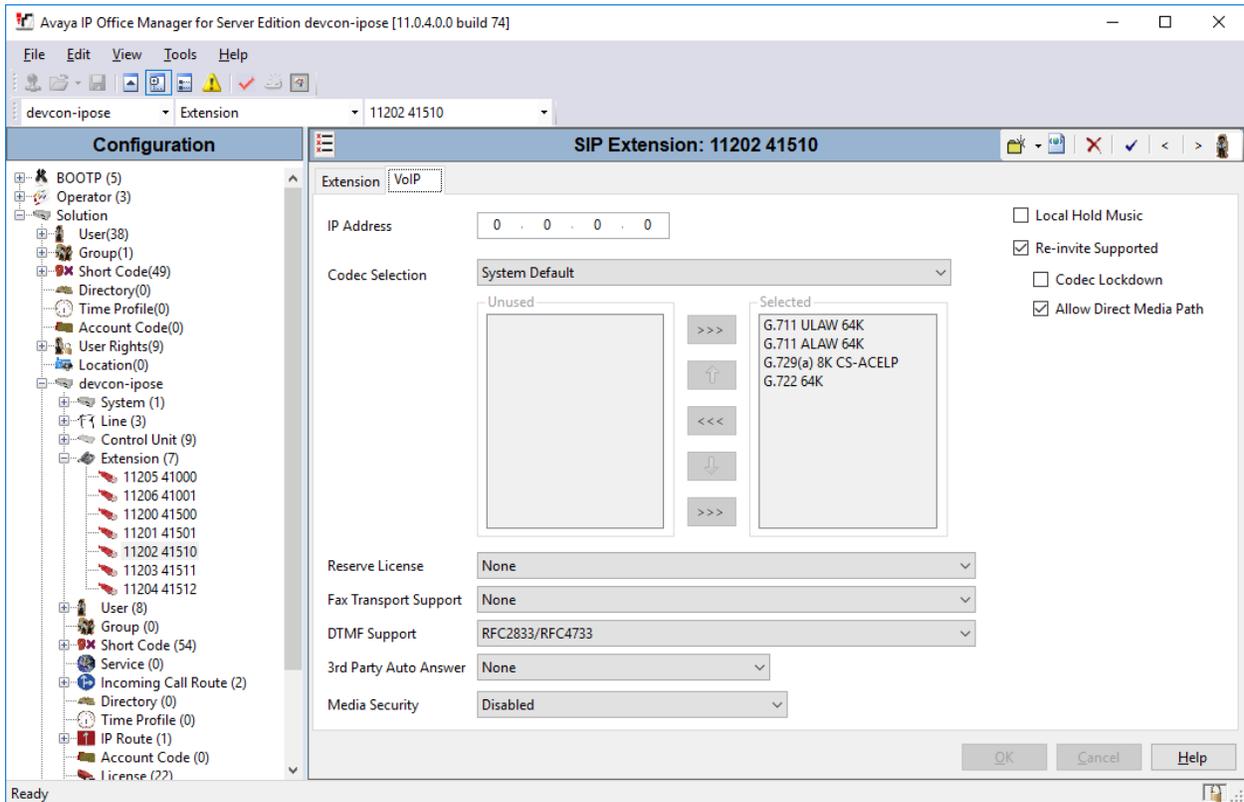


## 5.4. Administer SIP Extension

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, Talkphone IP Call Station was assigned extension *41510*.



Select the **VoIP** tab and retain the default values in the all fields. Talkphone IP Call Station supports G.711 and G.729 codecs.



## 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. 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 Talkphone IP Call Station to register with IP Office Server Edition. Note that the SIP authentication password was configured in the **Telephony** tab below instead of in the **Password** field shown below.

The screenshot displays the Avaya IP Office Manager for Server Edition interface. The left pane shows a configuration tree with 'User' selected under 'Solution'. The right pane shows the configuration for 'Talkphone41510: 41510' with the 'User' tab active. The configuration fields are as follows:

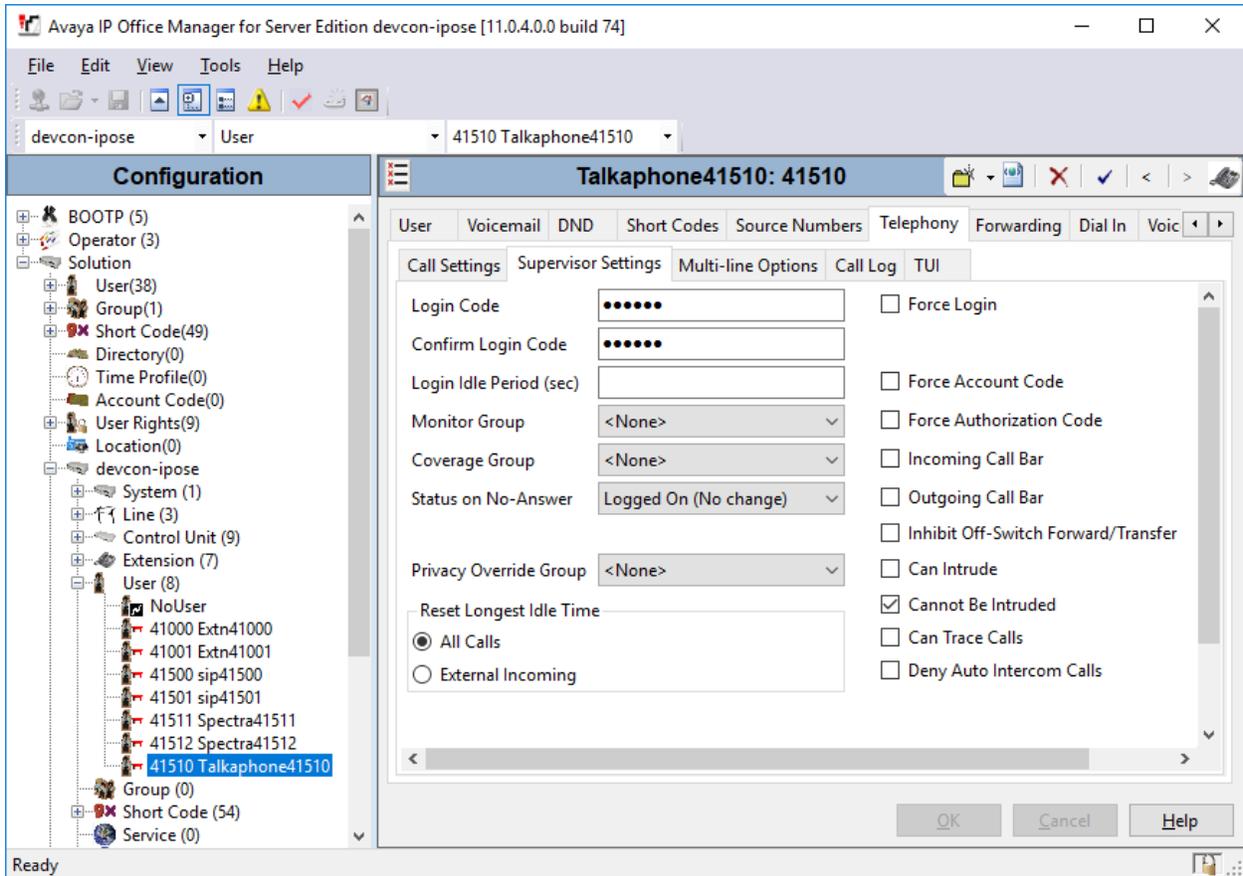
Field	Value
Name	Talkphone41510
Password	
Confirm Password	
Unique Identity	
Conference PIN	
Confirm Audio Conference PIN	
Account Status	Enabled
Full Name	Talkphone
Extension	41510
Email Address	
Locale	
Priority	5
System Phone Rights	None
Profile	Basic User

Below the profile dropdown, several checkboxes are visible, all of which are currently unchecked:

- Receptionist
- Enable Softphone
- Enable one-X Portal Services
- Enable one-X TeleCommuter
- Enable Remote Worker
- Enable Desktop/Tablet VoIP client
- Enable Mobile VoIP Client
- Send Mobility Email
- Web Collaboration

The interface includes a menu bar (File, Edit, View, Tools, Help), a toolbar, and a status bar at the bottom showing 'Ready'.

Select the **Supervisor Settings** sub-tab and enter a desired **Login Code**. The **Login Code** is the password that will be used by Talkphone IP Call Station to register with IP Office Server Edition.



## 6. Configure Talkphone VOIP-200 Series IP Call Station

This section covers the configuration of the Talkphone VOIP-200 Series IP Call Station. The following procedures are covered:

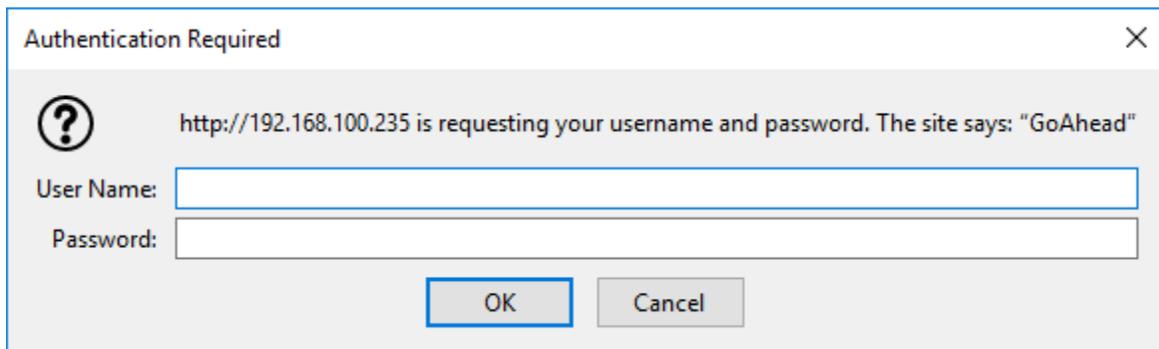
1. Launching the Web Administration Interface
2. Network Configuration
3. SIP Configuration
4. Configure Buttons

### 6.1. Launching the Web Administration Interface

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

- **IP Address:** 192.168.1.10
- **Username:** admin
- **Password:** admin@123

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



Authentication Required

ⓘ http://192.168.100.235 is requesting your username and password. The site says: "GoAhead"

User Name:

Password:

OK Cancel

## 6.2. Network Configuration

To modify the IP network configuration of the Talkphone IP Call Station, navigate to the **Configuration → IP Settings** page. Configure the IP settings so that it conforms to the customer network requirements. Click **Save** when done.

**TALKAPHONE**

Home Configuration Administration Diagnostics Network Security

IP Settings

DHCP  Static IP

IP Address: 192 . 168 . 100 . 235

Subnet Mask: 255 . 255 . 255 . 0

Gateway: 192 . 168 . 100 . 1

DNS Server 1: 0 . 0 . 0 . 0

DNS Server 2: 0 . 0 . 0 . 0

Hostname: voip000A01

Use last IP address on DHCP failure:

IGMP Version: Default

Save

## 6.3. SIP Configuration

Navigate to **Configuration → SIP Settings** to configure the SIP setting of the Talkphone IP Call Station. Configure the following parameters.

Under **Registration Settings**:

- **Display Name:** Specify a display name (e.g., *41510*).
- **Directory Number (SIP ID):** Specify the SIP number (e.g., *41510*) configured in **Section 5.5**.
- **Primary SIP Server:** Specify the IP address of IP Office Server Edition. (e.g., *10.64.102.90*).
- **Username:** Specify the SIP number of the Talkphone IP Call Station (e.g., *41510*).
- **Password:** Specify the SIP password configured in **Section 5.5**.

- **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., 5060).

Accept the default values for the **Call Settings** section and the remaining fields and click **Save** when done.

The screenshot shows the TALKAPHONE web interface. The navigation menu includes Home, Configuration, Administration, Diagnostics, and Network Security. The left sidebar contains a tree view with categories like IP Settings, SIP Settings, Audio Settings, Buttons, Auxiliary Output, Digital Outputs Scripts, Digital Outputs Events, Voice Messages Played to User, Voice Messages Played to Remote Side, and Time Settings. The main content area is divided into two sections: Registration Settings and Call Settings.

**Registration Settings**

Description	Configuration
Display Name:	41510
Directory Number (SIP ID):	41510
Primary SIP Server:	10.64.102.90
Secondary SIP Server:	
Tertiary SIP Server:	
Registration Method:	Parallel
Username:	41510
Password:	••••••
Re-registration Time:	3600 (Range: 60-14400 seconds)
Outbound Proxy 1 (optional):	10.64.102.90 Port: 5060
Outbound Proxy 2 (optional):	Port: 5060
Outbound Proxy 3 (optional):	Port: 5060

**Call Settings**

Description	Configuration
Enable auto-answer:	<input checked="" type="checkbox"/>
Auto-answer Delay:	0 seconds (Range: 0 to 30 seconds)
Provisional Timer:	0 seconds (Range: 0 to 60 seconds) Delays call setup using input buttons
Overlap dialing:	<input type="checkbox"/>
DTMF method:	RFC 2833
Call LED off during ringing:	<input type="checkbox"/>
Hang-up on Silence Timer:	0 seconds (0 = Hang-up on Silence disabled)
Codec G.711 PCM u-law:	High Priority
Codec G.711 PCM A-law:	Low Priority
Codec G.722:	Low Priority
Codec G.729:	Low Priority

At the bottom of the form is a "Save" button.

## 6.4. Configure Buttons

Navigate to **Configuration** → **Buttons** to verify the appropriate settings. For the compliance test, the **Buttons** were configured as shown below. The **Value** field for **Button 1** was set to a valid extension. This is the destination that will be dialed when the call button is pressed. In the **Buttons (Active Call)** section, set the **Button 1 Function** field to Disconnect. This will allow an active call to be disconnected when the call button is pressed. This is optional based on customer requirements.

The screenshot shows the TALKAPHONE configuration interface. The navigation menu includes Home, Configuration, Administration, Diagnostics, and Network Security. The left sidebar contains various settings categories, with 'Buttons' selected. The main content area is divided into three sections: Buttons (Idle), Buttons (Active Call), and Numberlist Settings.

**Buttons (Idle)**

	Function	Value	Option
Button 1	Call To	41001	None
Button 2	Call To		None
Button 3	Call To		None

**Buttons (Active Call)**

	Function	Activated	Deactivated
Button 1	Disconnect	<input type="checkbox"/>	<input type="checkbox"/>
Button 2	Do Nothing	<input type="checkbox"/>	<input type="checkbox"/>
Button 3	Do Nothing	<input type="checkbox"/>	<input type="checkbox"/>

**Numberlist Settings**

	Ringlist 1	With Previous	Ringlist 2	With Previous	Ringlist 3	With Previous
Value 1	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 2	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 3	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 4	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>

Scroll down to the **Numberlist Settings** section to view and customize call parameters, such as **Ringer Time**, which may need to be increased to provide enough time for the destination to cover to an alternate destination, if necessary, **Local Interdigit Timer**, which dictates how long to wait before initiating a call after the user dials the digits, or the **Call Conversation Timer**, which specifies how long an emergency call should remain active, unless the far-end drops the call. The following screen shows the default values for the call parameters. Click **Save** when done.

**Note:** After a number is dialed on the Talkphone IP Call Station, the **Local Interdigit Timer** must expire before the call is initiated. The minimum value for the **Local Interdigit Timer** is 5 secs.

Numberlist Settings						
	Ringlist 1	With Previous	Ringlist 2	With Previous	Ringlist 3	With Previous
Value 1	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 2	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 3	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 4	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 5	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 6	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 7	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 8	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 9	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 10	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 11	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 12	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 13	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 14	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Call Until Answer	<input checked="" type="checkbox"/> (loops the numberlist)					
Ringer Time	<input type="text" value="10"/> seconds, (0=unlimited)					
Call Conversation Timer	<input type="text" value="720"/> seconds, (Range: 0 to 9999 seconds, 0 = unlimited)					
Local Interdigit Timer	<input type="text" value="5"/> seconds, (Range: 5 to 20 seconds)					
<input type="button" value="Save"/>						

## 7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Talkphone VOIP-200 Series IP Call Stations with Avaya IP Office Server Edition.

1. Verify that Talkphone IP Call Station has successfully registered with IP Office Server Edition. Launch **IP Office System Status** and navigate to **Extensions** → *<SIP Extension>*, where *<SIP Extension>* is the IP Call Station extension. Verify that the **Current State** is *Idle* as shown below.

The screenshot displays the Avaya IP Office System Status application window. The title bar reads "Avaya IP Office System Status - devcon-ipose (10.64.102.90) - IP Office Linux PC 11.0.4.0.0 build 74". The main window has a blue header with the Avaya logo and the title "IP Office System Status". Below the header is a menu bar with "Help", "Snapshot", "LogOff", "Exit", and "About".

The left sidebar contains a tree view with the following items: System, Alarms (5), Extensions (4) (with sub-items 41000, 41001, 41501, and 41510 selected), Trunks (3), Active Calls, Resources, Voicemail, IP Networking, and Locations.

The main content area is titled "Extension Status" and displays the following details for extension 41510:

- Extension Number: 41510
- IP address: 192.168.100.236
- Standard Location: None
- Registrar: Primary
- Telephone Type: Unknown SIP Device
- User-Agent SIP header: Zenitel IPSTATION v2.0
- Media Stream: RTP
- Layer 4 Protocol: UDP
- Current User Extension Number: 41510
- Current User Name: Talkphone41510
- Forwarding: Off
- Twinning: Off
- Do Not Disturb: Off
- Message Waiting: Off
- Phone Manager Type: None
- SIP Device Features: REFER
- License Reserved: No
- Last Date and Time License Allocated: 4/1/2019 11:39:02 AM
- Packet Loss Fraction: [blank]
- Jitter: [blank]
- Round Trip Delay: [blank]
- Connection Type: [blank]
- Codec: [blank]
- Remote Media Address: [blank]

Below the details is a table with the following columns: Call Ref, Current State, Time in State, Calling Number or Called Number, Direction, and Other Party on Call. The table contains one row with the following data:

Call Ref	Current State	Time in State	Calling Number or Called Number	Direction	Other Party on Call
	Idle	02:40:32			

At the bottom of the main content area are several buttons: Trace, Trace All, Pause, Ping, Call Details, Print..., and Save As... The bottom status bar shows the time "2:19:35 PM" and the status "Online".

2. Place an incoming/outgoing call to to/from the Talkphone IP Call Station, verify 2-way audio and proper call termination.

## 8. Conclusion

These Application Notes have described the administration steps required to integrate the Talkphone VOIP-200 Series IP Call Stations with Avaya IP Office Server Edition. Talkphone 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 Platform with Manager*, Release 11.0 FP4, February 2019, available at <http://support.avaya.com>.

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