



## DevConnect Program

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# Application Notes for Talkphone VOIP-500 Series IP Call Stations with Avaya IP Office Server Edition 11.1- Issue 1.0

## Abstract

These Application Notes describe the configuration steps required to integrate the Talkphone VOIP-500 Series IP Call Stations 1.0.3.2 with Avaya IP Office Server Edition 11.1 and Avaya IP Office 500V2 Expansion System 11.1. Talkphone VOIP-500 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-500 Series IP Call Stations register with Avaya IP Office as a SIP endpoint. For the compliance test, a Talkphone VOIP-500-ECK IP Call Station was used, which provides a call button, an emergency call button, and keypad. Talkphone VOIP-500ECK IP call station does not have a handset.

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 Avaya DevConnect Program.

# 1. Introduction

These Application Notes describe the configuration steps required to integrate the Talkaphone VOIP-500 Series IP Call Stations 1.0.3.2 with Avaya IP Office Server Edition 11.1 and Avaya IP Office 500V2 Expansion System 11.1. Talkaphone VOIP-500 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-500 Series IP Call Stations support SIP (RFC 3261) and can operate as a paging/mass notification device via a standard SIP-based inbound call. Talkaphone VOIP-500 Series IP Call Stations register with Avaya IP Office as a SIP endpoint. For the compliance test, a Talkaphone VOIP-500ECK IP Call Station was used, which provides a call button, an emergency call button, and keypad. Talkaphone VOIP-500ECK IP call station does not have a handset.

## 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-500-ECK IP Call Station, 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-500ECK IP Call Station came back into service after re-connecting the Ethernet cable or rebooting the VOIP-500ECK 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-500ECK IP Call Station did not include use of any specific encryption features as requested by Talkaphone.

## 2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of VOIP-500-ECK with IP Office Server Edition or IP Office 500V2 Expansion.
- Inbound and outbound calls between VOIP-500ECK 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.
- G.711 and G.729 codec support.
- Calls between VOIP-500ECK and the PSTN.
- Basic telephony features, including hold, mute, redial, call forwarding, transfer, and 3-way conference, initiated from the perspective of Avaya IP deskphones.
- Use of paging, recorded messages, emergency calls, and number lists on the VOIP-500ECK IP call station.
- Proper system recovery after a restart of VOIP-500ECK Station and loss of IP connectivity.

## 2.2. Test Results

All test cases passed with the following observation:

- Emergency calls cannot be terminated from the Talkphone VOIP-500ECK IP Call Station. The calls can only be disconnected by the called party or upon expiration of the “Call Conversation Timer” set in the Talkphone VOIP-500ECK. The Talkphone VOIP-500ECK IP Call Station has an emergency button that when pressed dials a list of programmed numbers in a round-robin fashion. If the first number in the list does not answer (i.e., Busy, No Answer, Out of Order, Invalid number), 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. The programmed numbers should not cover to voicemail, or the user will end up on a call with the voicemail system without being able to disconnect.
- Talkphone DTMF tones sending, welcome, paging, and recording features may not work when Direct IP Media (shuffling) is enabled.
- Dialing short codes to activate telephony features are not applicable to Talkphone IP Call Stations.

## 2.3. Support

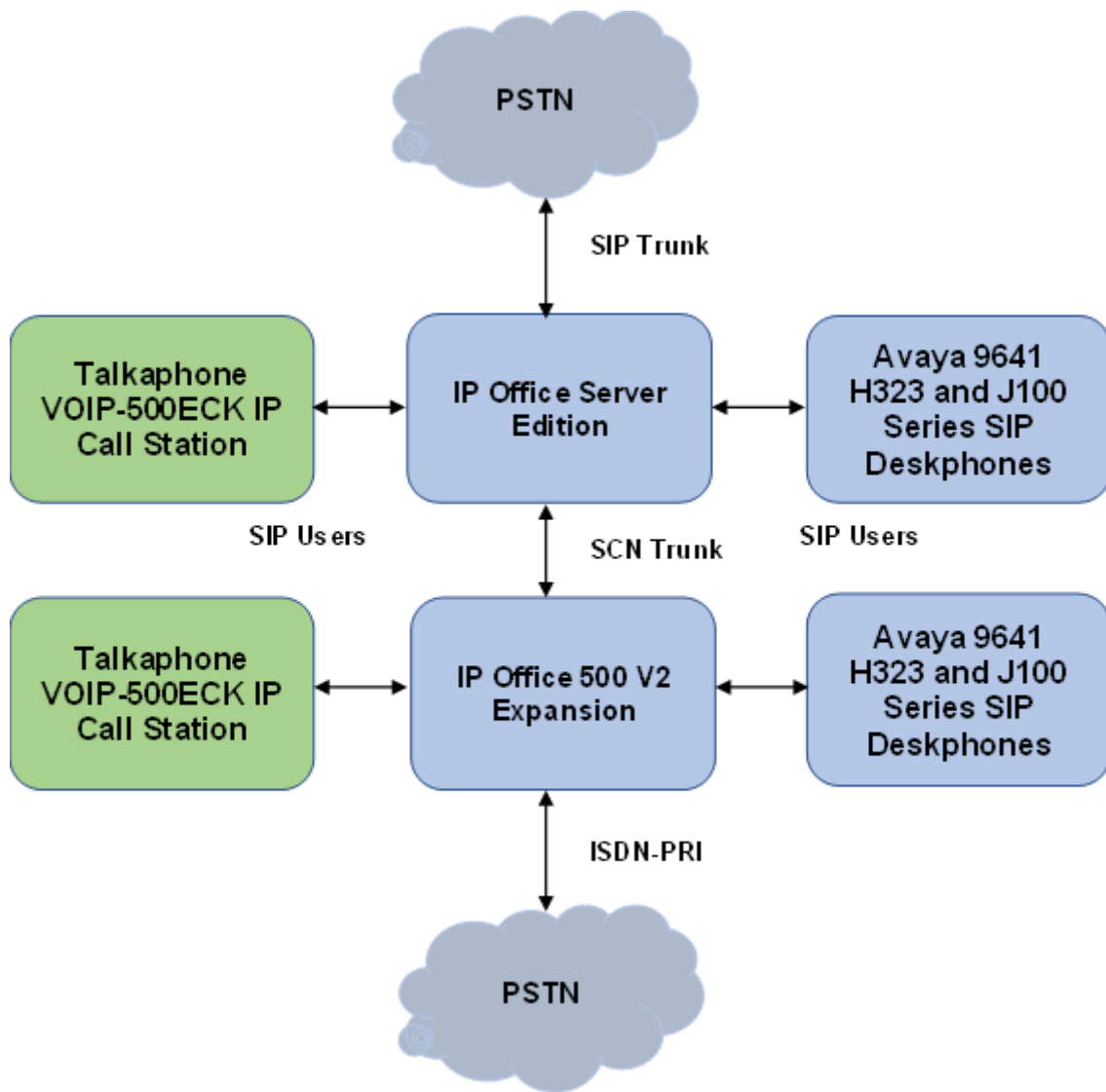
For technical support and information on Talkaphone VOIP-500 Series IP Call Stations, contact Talkaphone Technical Support at:

- Phone: 1-773-539-1100
- Email: [support@talkaphone.com](mailto: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.
- Talkphone VOIP-500ECK 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 Talkphone VOIP-500ECK 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	11.1.3.0.0 build 23
Avaya IP Office 500 V2 Expansion	11.1.3.0.0 build 23
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-500ECK IP Call Station	1.0.3.2

**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:

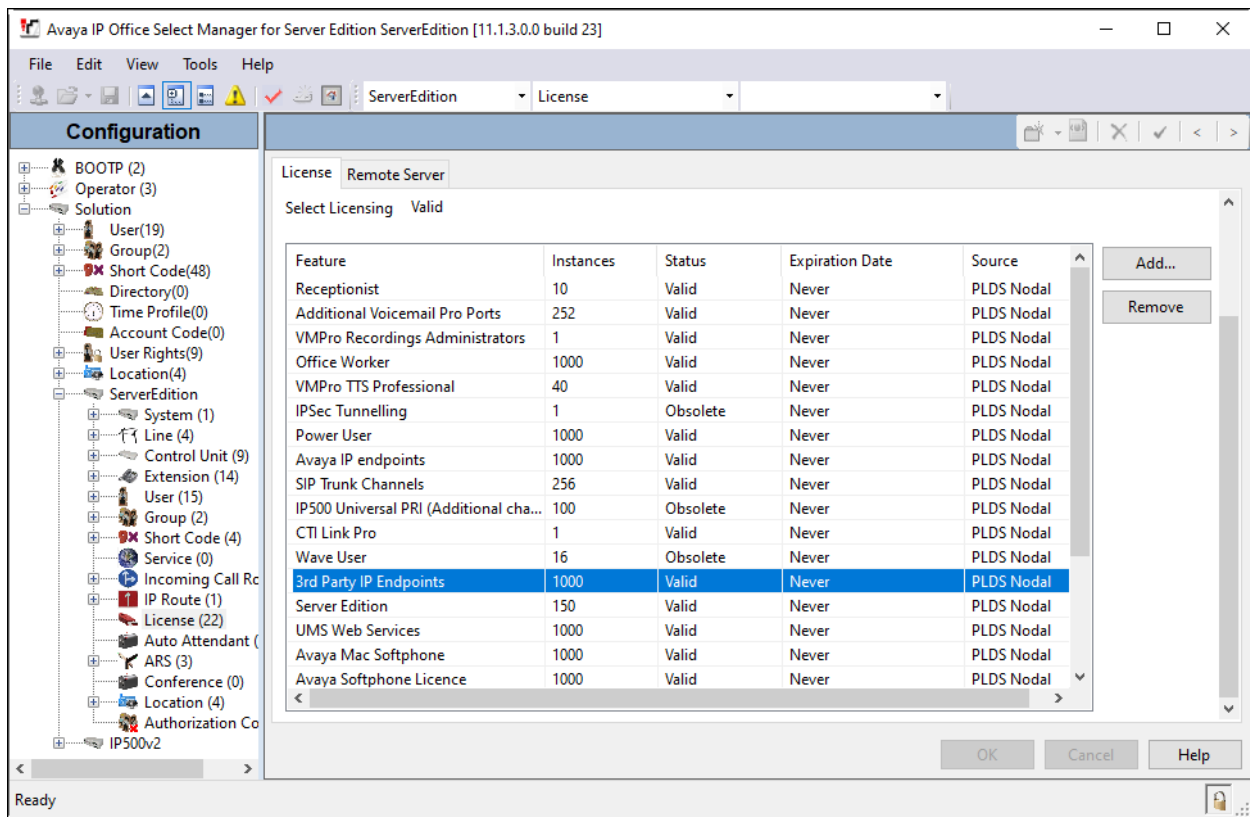
- Verify IP Office License
- Obtain LAN IP Address
- Administer SIP Registrar
- Administer SIP Extension
- Administer SIP User

**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.

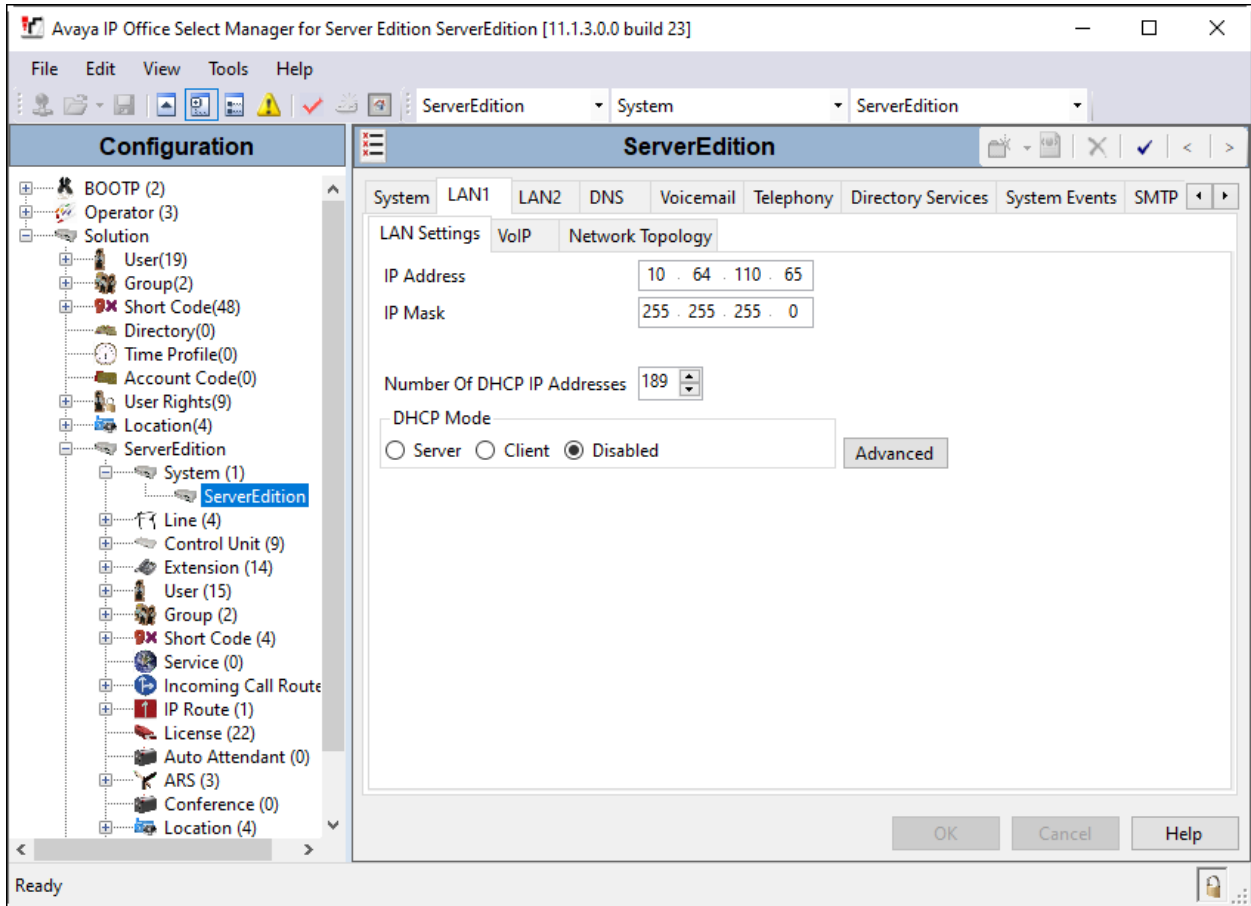
The **Avaya IP Office Manager for Server Edition** screen is displayed. From the configuration tree in the left pane, select **License** to display the license screen in the right pane. Verify that the **License Status** is “Valid” for **3rd Party IP Endpoints**.



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 Tunnelling	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

## 5.2. Obtain LAN IP Address

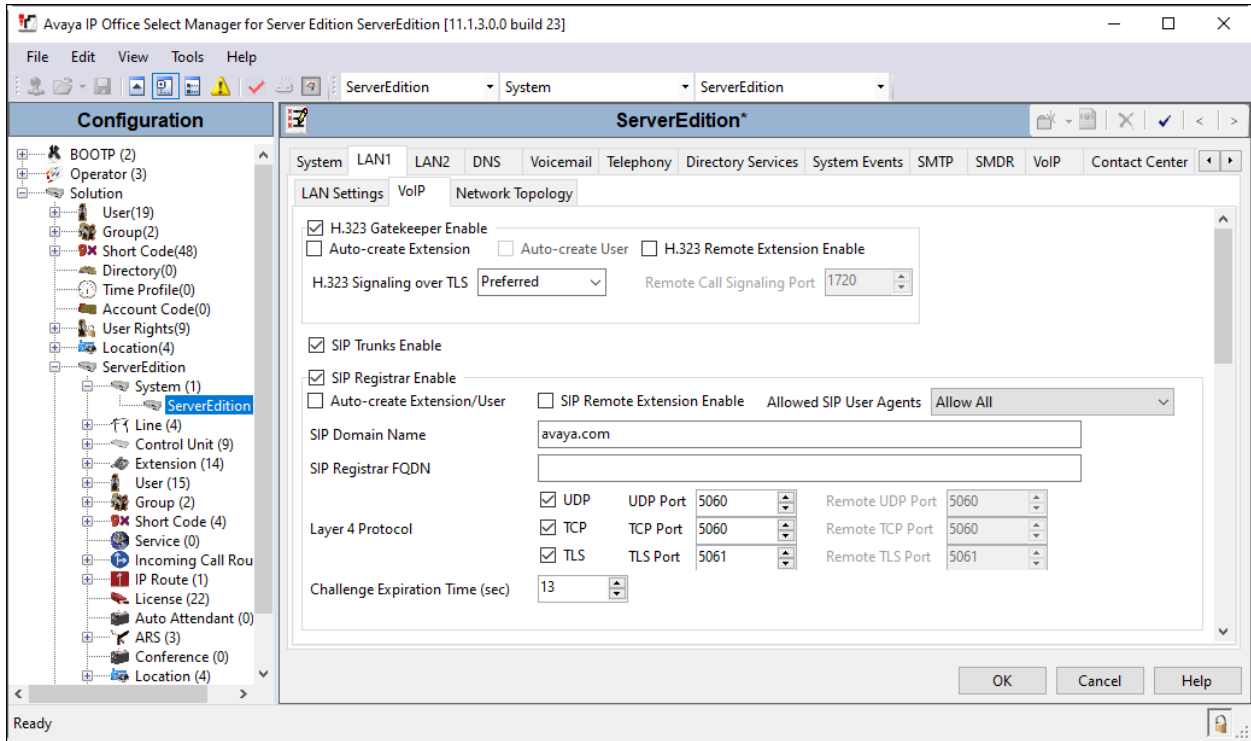
From the configuration tree in the left pane, select **System** to display the **System** screen for the IP Office Server Edition in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure VOIP-500ECK.





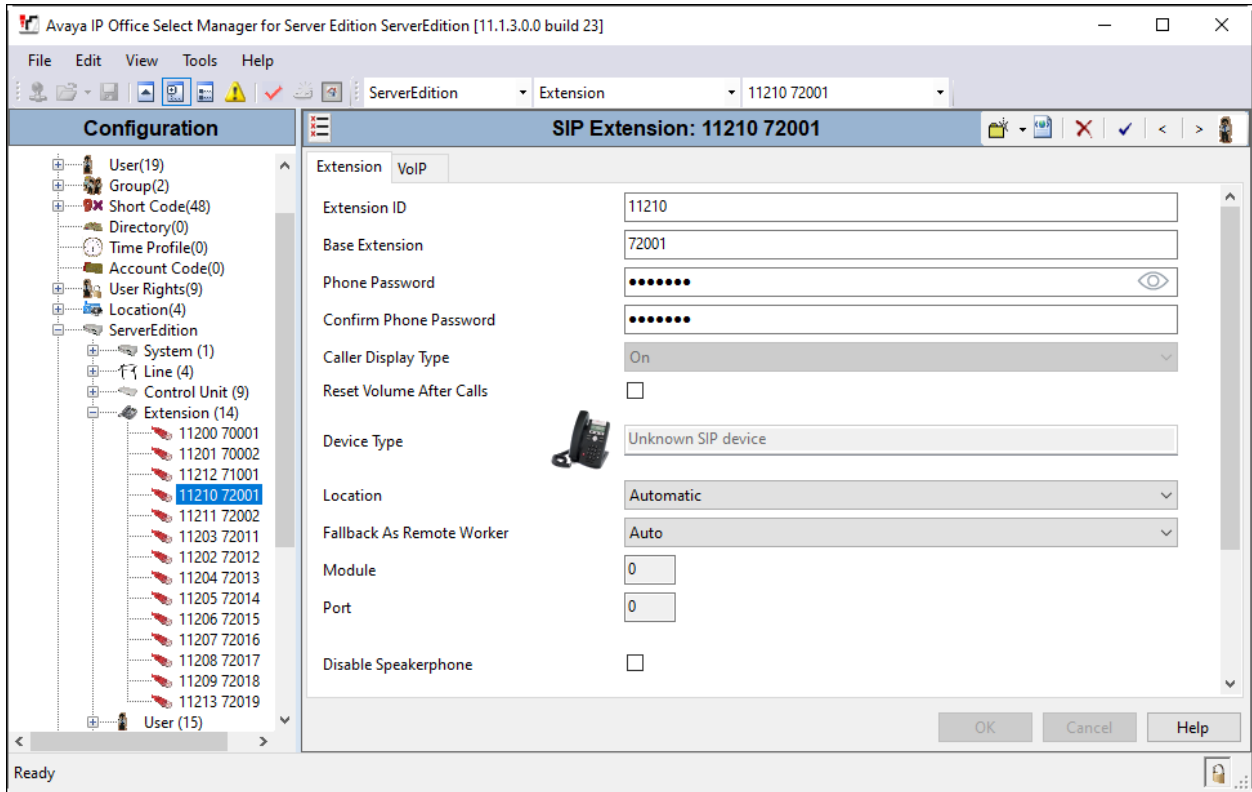
### 5.3. Administer SIP Registrar

Select the **VoIP** sub-tab. Ensure that **SIP Registrar Enable** and that UDP transport is selected, which will be used by VOIP-500ECK, and enter a valid **SIP Domain Name** (e.g., *avaya.com*).

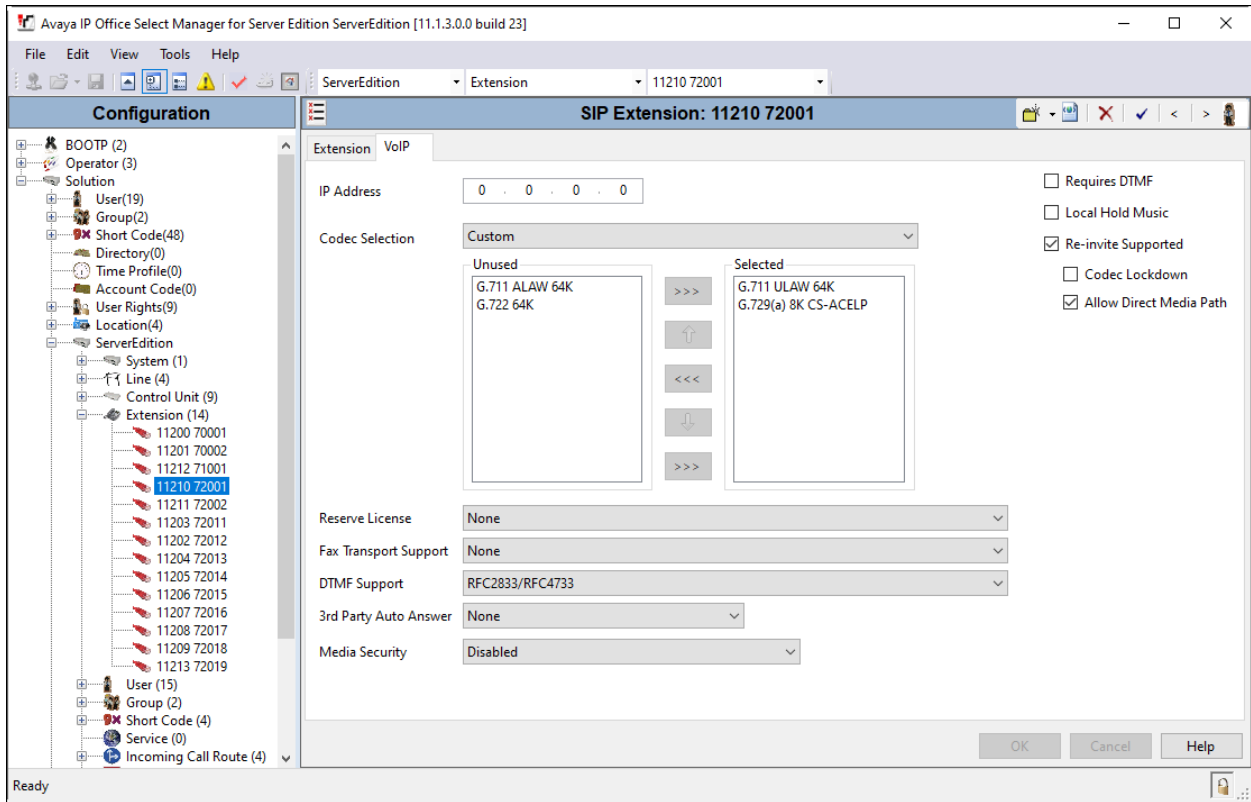


## 5.4. Administer SIP Extension

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-500ECK was assigned extension 72001. Configure the **Phone Password** that will be used by VOIP-500ECK to register with IP Office Server Edition.

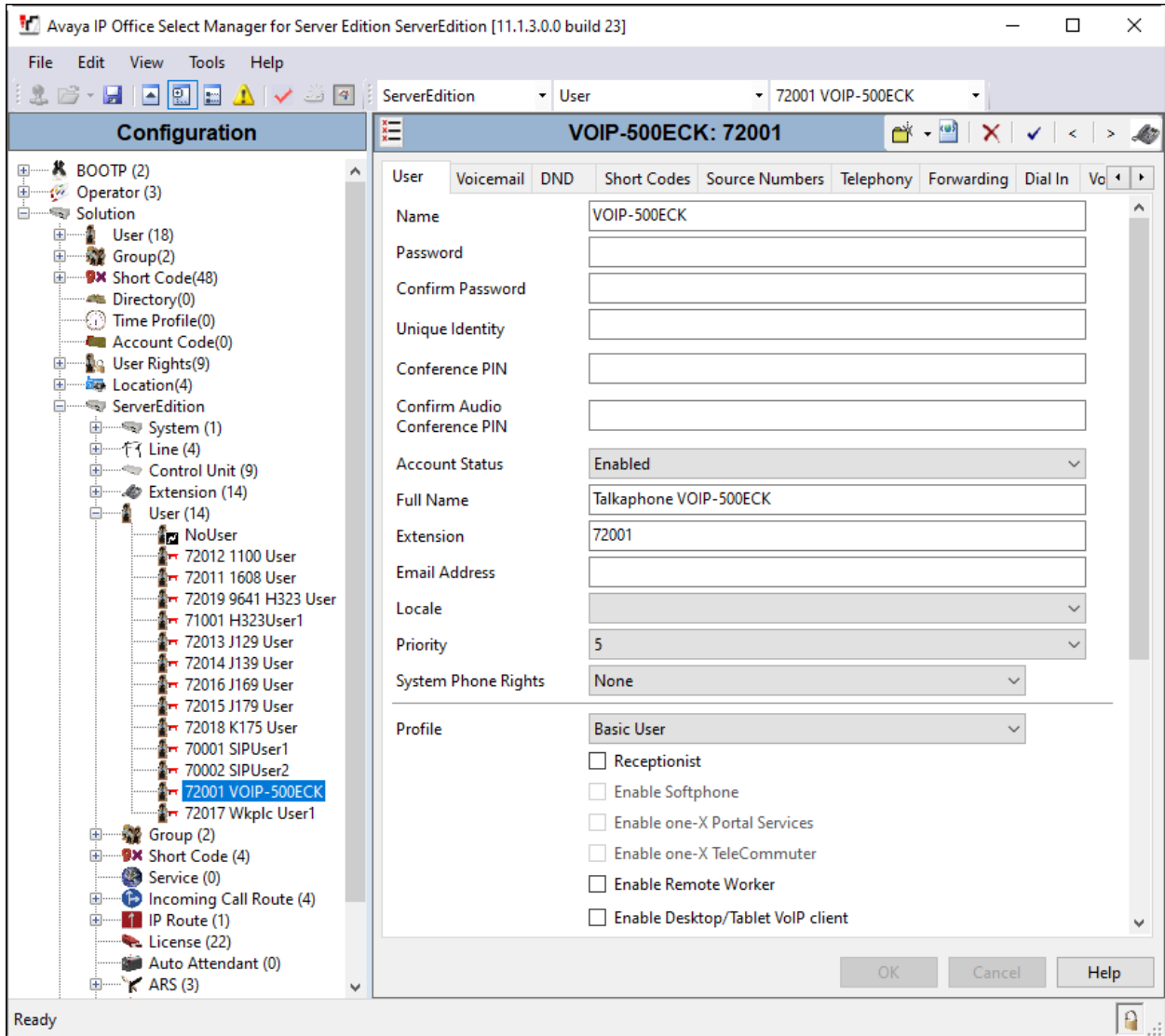


Select the **VoIP** tab. For **Codec Selection**, supported codecs may be selected and can be customized to the extension if desired. For the compliance test, G.711MU and G.729 were verified with VOIP-500ECK. **Media Security** was disabled for VOIP-500ECK.

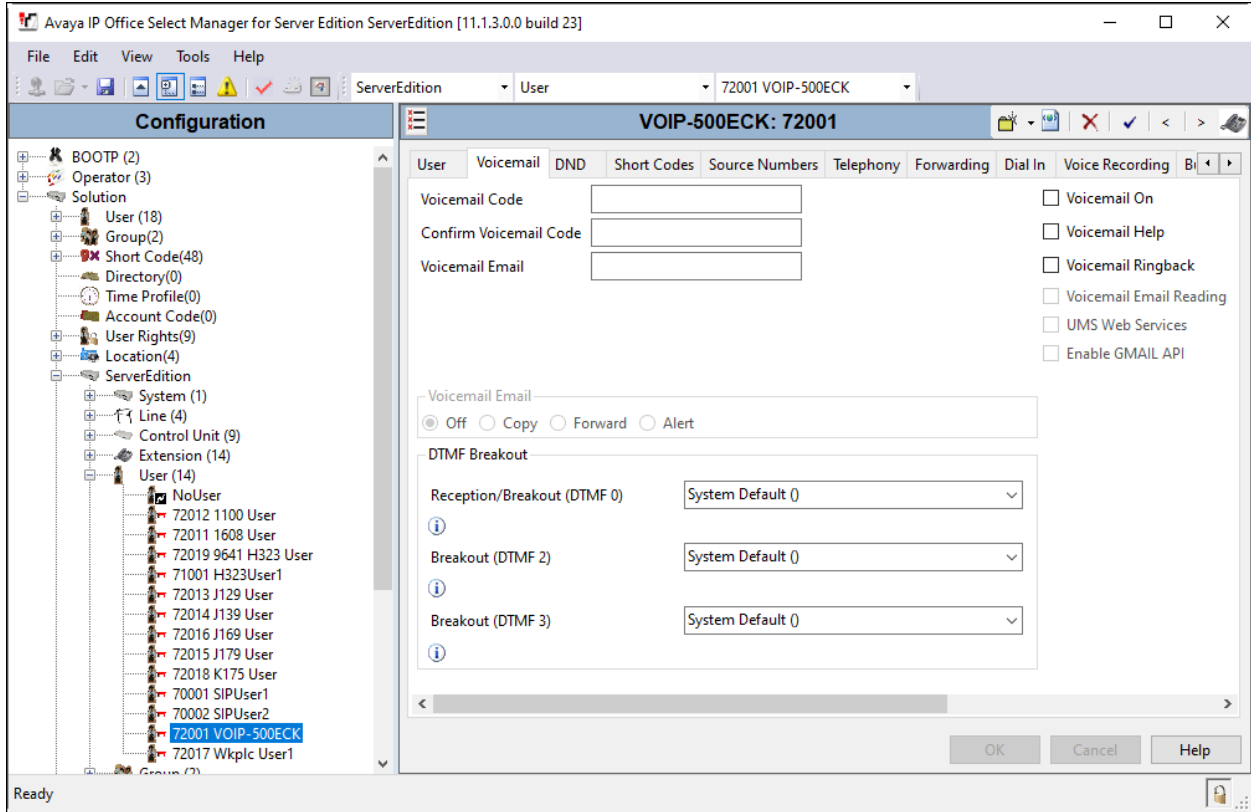


## 5.5. Administer SIP User

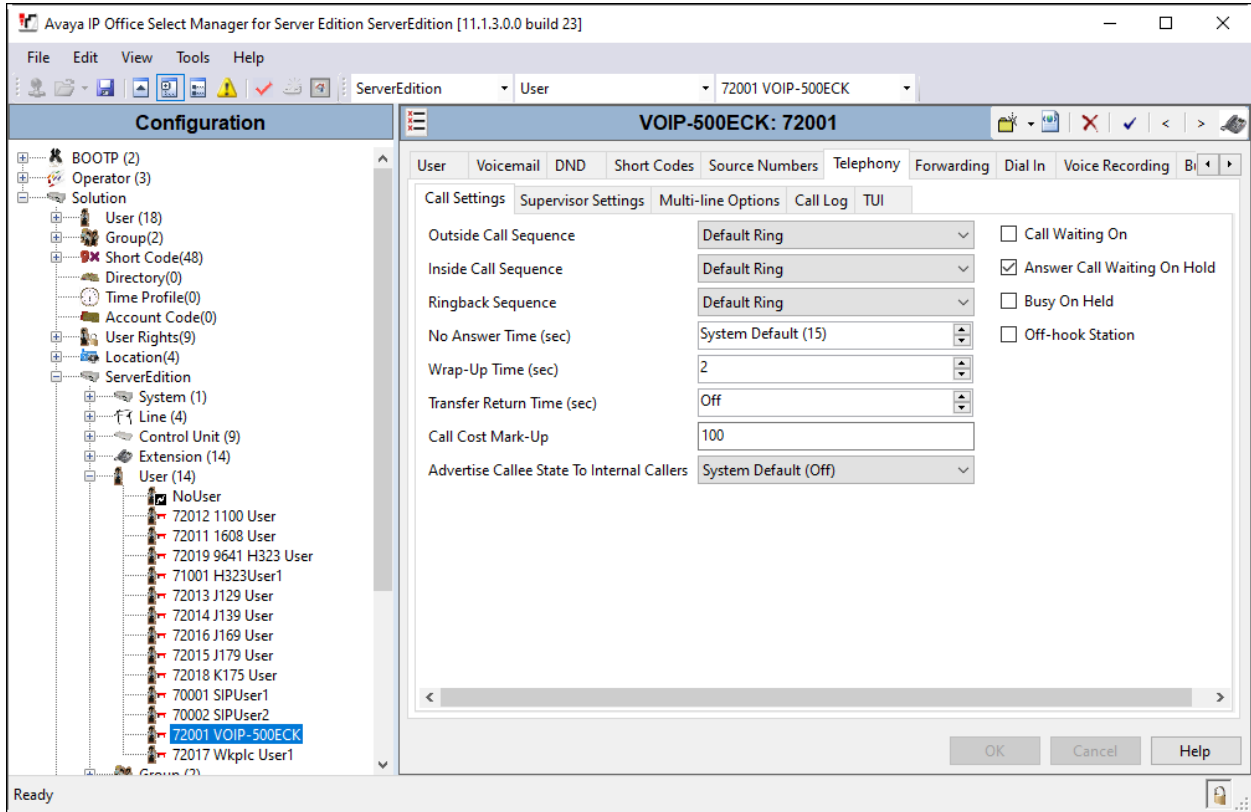
From the configuration tree in the left pane, right-click on **User** and select **New** from the pop-up list (not shown). Enter desired values for the **Name** and **Full Name** fields. For the **Extension** field, enter the SIP extension created in **Section 5.4**. The **Extension** field specifies the username that will be used by VOIP-500ECK to register with IP Office Server Edition.



Select the **Voicemail** tab and disable voicemail for VOIP-500ECK.



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



## 6. Configure Talkphone VOIP-500ECK IP Call Station

This section covers the configuration of the Talkphone VOIP-500ECK IP Call Station. The following procedures are covered:

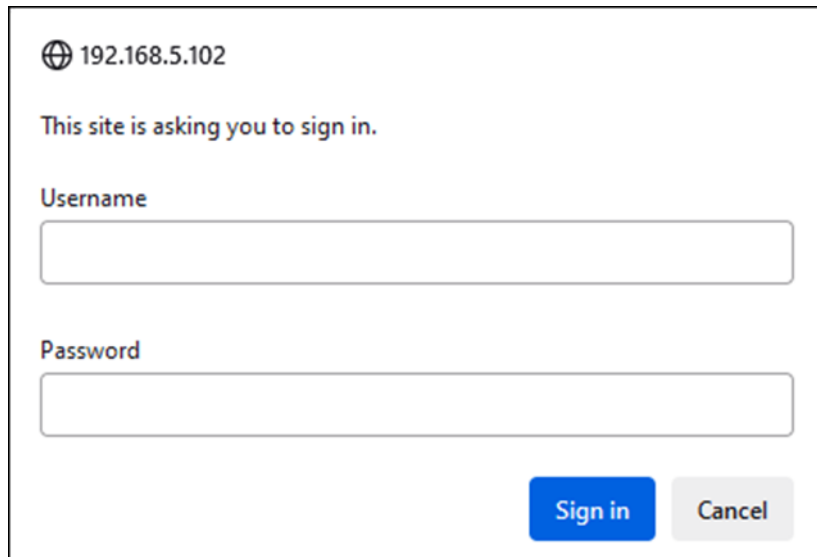
- Launching the Web Administration Interface
- Network Configuration
- SIP Configuration
- Configure Audio Settings
- Configure Call Parameters
- Configure Buttons
- Configure Number Lists

### 6.1. Launching the Web Administration Interface

Talkphone IP VOIP-500ECK Call Stations are pre-configured with the following default values:

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

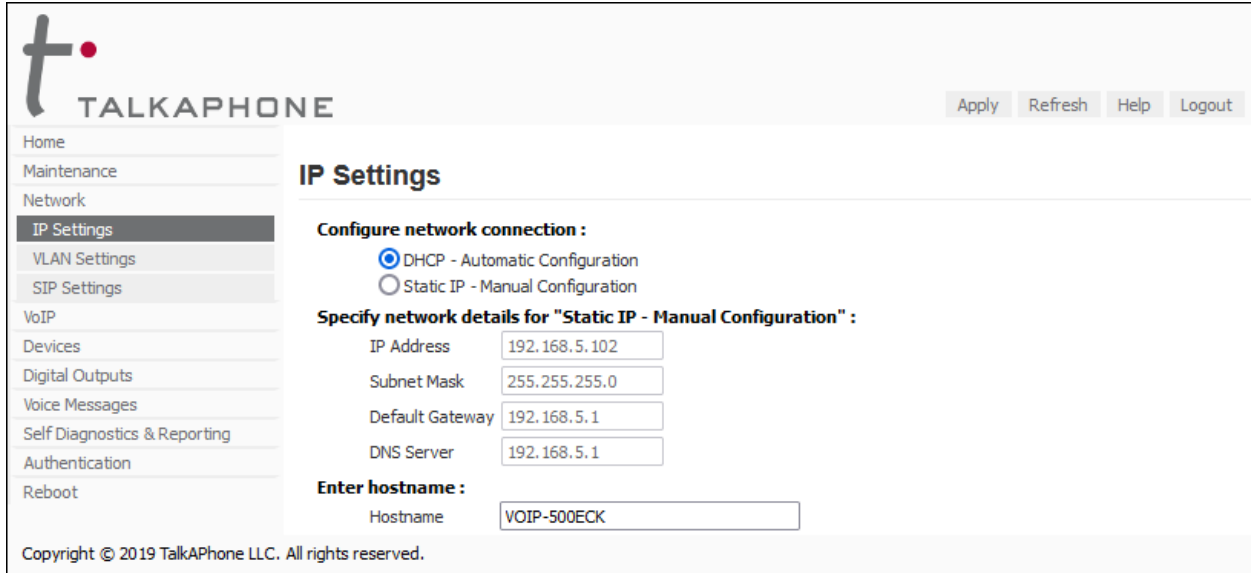
Ensure that the administration PC and Talkphone VOIP-500ECK IP Call Station are connected to the LAN. Open a web browser (Talkphone supports Mozilla 3.5 and above) and enter the default IP address of the Talkphone VOIP-500ECK IP Call Station in the URL field. The browser prompts for authentication. Log in with the appropriate credentials.



The screenshot shows a web browser's authentication dialog box. At the top, it displays a globe icon followed by the IP address "192.168.5.102". Below this, the text "This site is asking you to sign in." is centered. There are two input fields: the first is labeled "Username" and the second is labeled "Password". At the bottom right, there are two buttons: a blue "Sign in" button and a grey "Cancel" button.

## 6.2. Network Configuration

To modify the IP network configuration of the Talkaphone VOIP-500ECK IP Call Station, navigate to the **Network → IP Settings** page. Configure the IP settings so that it conforms to the customer network requirements. Compliance testing used *DHCP* and a **Hostname** of *VOIP-500ECK*. Click **Apply** when done.



The screenshot shows the Talkaphone web interface for IP Settings. The page title is "IP Settings". On the left is a navigation menu with options: Home, Maintenance, Network (selected), IP Settings (selected), VLAN Settings, SIP Settings, VoIP, Devices, Digital Outputs, Voice Messages, Self Diagnostics & Reporting, Authentication, and Reboot. The main content area is titled "IP Settings" and contains the following configuration options:

- Configure network connection :**
  - DHCP - Automatic Configuration
  - Static IP - Manual Configuration
- Specify network details for "Static IP - Manual Configuration" :**
  - IP Address: 192.168.5.102
  - Subnet Mask: 255.255.255.0
  - Default Gateway: 192.168.5.1
  - DNS Server: 192.168.5.1
- Enter hostname :**
  - Hostname: VOIP-500ECK

At the top right of the interface are buttons for "Apply", "Refresh", "Help", and "Logout". At the bottom left, the copyright notice reads: "Copyright © 2019 TalkAPhone LLC. All rights reserved."

## 6.3. SIP Configuration

Navigate to **Network → SIP Settings** to configure the SIP setting of the Talkaphone IP VOIP-500ECK Call Station. Configure the following parameters.

Under **Assign a phone number:**

- **Phone Number:** Specify the extension used (e.g., *72001*)

Under **Specify SIP Server FQDN/IP Address:**

- **Primary SIP Server FQDN/IP Address:** Specify the IP address of IP Office Server Edition (e.g., *10.64.110.65*).

Under **Specify Outbound Proxy:**

- **Enable/ Disable SIP registration:** Select this item.



Under **Specify SIP registrar** and **Specify outbound proxy**:

- **Username:** Specify the SIP number of the Talkphone IP Call Station (e.g., 72001).
- **Password:** Specify the SIP password configured in **Section 5.4**.
- **Primary SIP Registrar:** Specify the IP address of IP Office Server Edition (e.g., 10.64.110.65)
- **Outbound Proxy 1 IP Address:** Specify the IP address of IP Office Server Edition (e.g., 10.64.110.65).
- **Port:** Specify the SIP port (e.g., 5060).

Accept the default values for the remaining fields and click **Apply** when done.

**TALKAPHONE**

Apply Refresh Help Logout

Home  
Maintenance  
Network  
IP Settings  
VLAN Settings  
**SIP Settings**  
VoIP  
Devices  
Digital Outputs  
Voice Messages  
Self Diagnostics & Reporting  
Authentication  
Reboot

### SIP Settings

**Assign a phone number :**

Phone Number

**Specify SIP Server FQDN/IP Address :**

Primary SIP Server FQDN/IP Address

Secondary SIP Server FQDN/IP Address

Tertiary SIP Server FQDN/IP Address

**Enable / disable SIP registration :**

Register

**Specify SIP registrar :**

Username

Password

Primary SIP Server IP Address

Secondary SIP Server IP Address

Tertiary SIP Server IP Address

Port  (Port Range: 1024-49151)

Re-registration Time  (Range: 10-14400 seconds)

**Specify outbound proxy :**

Username

Password

Outbound Proxy 1 IP Address

Outbound Proxy 2 IP Address

Outbound Proxy 3 IP Address

Port  (Port Range: 1024-49151)

## 6.4. Configure Audio Settings

Navigate to **VoIP → Audio Settings** to configure the preferred codec and microphone and speaker parameters. In addition, the Speaker Gain can be adjusted to control the volume. All other fields were left at the default values. Click **Apply** when done.

The screenshot shows the 'Audio Settings' page in the TalkAPhone web interface. The page has a sidebar on the left with navigation links: Home, Maintenance, Network, VoIP (with sub-links for Number Lists, Phone Settings, Audio Settings, Call Parameters, and Paging Settings), Devices, Digital Outputs, Voice Messages, Self Diagnostics & Reporting, Authentication, and Reboot. The main content area is titled 'Audio Settings' and contains several sections:

- Select VoIP codec :** Radio buttons for G.711 PCM a-Law @ 64kbps, G.711 PCM u-Law @ 64kbps (selected), G.729a, and G.723.1a.
- Enable/disable audio processing modules :** Checkboxes for VAD/CNG (checked), AEC (checked), and AGC (unchecked). A 'Jitter Buffer' dropdown is set to 30 ms.
- DTMF duration for outgoing calls :** Radio buttons for Disable, 51 ms, 60 ms (selected), 102 ms, and Custom. A 'Duration' input field is set to 800 ms, with a note '(Range: 10-1000 ms)'. There is also a 'Duration' input field set to 800 ms.
- Configure Line Level Output parameters :** A 'Line Gain' dropdown is set to 16.
- Configure Speaker/Microphone parameters :** Checkboxes for Speaker (checked), Microphone (checked), and Use Speaker for notification and ringing only (unchecked). To the right, 'Speaker Gain' and 'Microphone Gain' dropdowns are both set to 12.

At the top right of the page are buttons for 'Apply', 'Refresh', 'Help', and 'Logout'. At the bottom left, the copyright notice reads: 'Copyright © 2019 TalkAPhone LLC. All rights reserved.'

## 6.5. Configure Call Parameters

Navigate to **VoIP → Call Parameters** to view and customize any of the call parameters, such as **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.

**Note:** After a number is dialed on the Talkaphone VOIP-500ECK 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.

**TALKAPHONE** Apply Refresh Help Logout

Home  
Maintenance  
Network  
VoIP  
Number Lists  
Phone Settings  
Audio Settings  
**Call Parameters**  
Paging Settings  
Devices  
Digital Outputs  
Voice Messages  
Self Diagnostics & Reporting  
Authentication  
Reboot

### Call Parameters ✓ Updated.

**Enable/disable call progress tones :**  
 Enable  Disable

**Specify key to answer and/or disconnect a call from the Remote Side :**  
To disconnect a call, press    
To answer a call, press

**Enable/disable "Welcome Tone" :**  
 Enable  Disable

**Configure required timers :**

Provisional Timer	<input type="text" value="5"/>	(Range: 5-20 seconds)
Ringer Timer	<input type="text" value="5"/>	(Range: 1-12 rings)
Hang-up Timer	<input type="text" value="0.5"/>	(Range: 0.5-3.0 seconds)
Local Interdigit Timer	<input type="text" value="5"/>	(Range: 5-20 seconds)
Remote Interdigit Timer	<input type="text" value="5"/>	(Range: 5-20 seconds)

**Configure optional timers :**

<input checked="" type="checkbox"/> Call conversation Timer	<input type="text" value="12"/>	(Range: 1-360 min.)
<input checked="" type="checkbox"/> Ringback or Busy Timer	<input type="text" value="15"/>	(Range: 1-60 seconds)
<input checked="" type="checkbox"/> Hang-up On Silence Timer	<input type="text" value="30"/>	(Range: 10-360 seconds)

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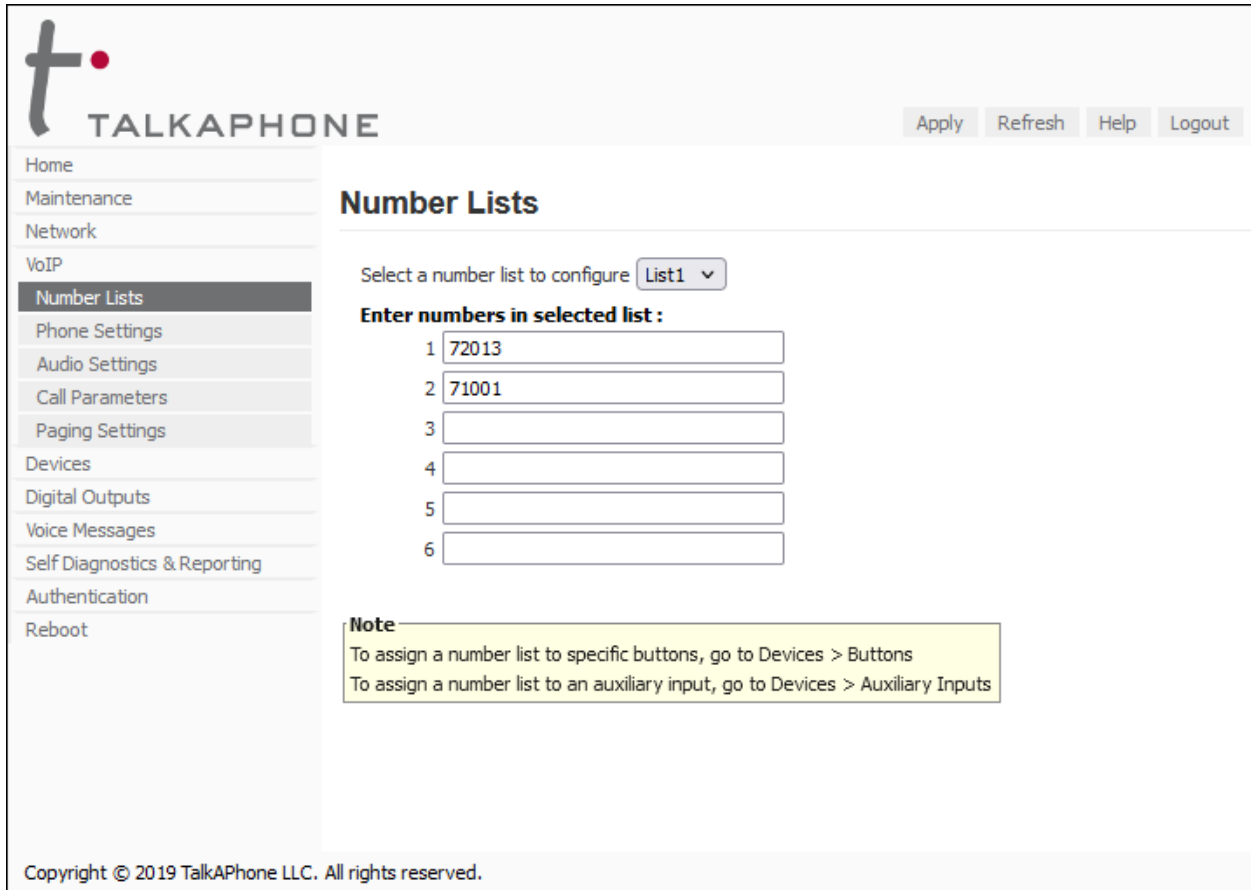
## 6.6. Configure Buttons

Navigate to **Devices** → **Buttons** to verify the appropriate settings. For the compliance test, the **Buttons** were configured as shown below. Note that the **Call from Number List** specifies the extension list that the Talkphone VOIP-500ECK IP Call Station will call when the emergency button is pressed. In this case, *List 1* is used.

The screenshot displays the TalkPhone web interface for configuring buttons. The sidebar menu on the left includes options like Home, Maintenance, Network, VoIP, Devices, and Buttons (which is currently selected). The main content area is titled 'Buttons' and contains two configuration sections: 'Configure Button #1' and 'Configure Button #2'. Each section has several settings: 'Button #1 Mode' is set to 'Always Autodial' with an unchecked checkbox; 'Call from Number List' is set to 'List 1'; 'Call Priority' is set to '1'; and 'Network Priority' is set to '46' with a range of 0-63. Similarly, 'Configure Button #2' has 'Button #2 Mode' set to 'Hook Switch', 'Call from Number List' set to 'List 1', 'Call Priority' set to '2', and 'Network Priority' set to '46' with a range of 0-63. At the bottom of the interface, there is a copyright notice: 'Copyright © 2019 TalkAPhone LLC. All rights reserved.'

## 6.7. Configure Number Lists

Navigate to **VoIP → Number Lists** to specify the numbers to be called when the emergency button is pressed. In this case, extension 72013 is dialed first. If there is no answer, then extension 71001 is dialed. Talkaphone VOIP-500ECK IP Call Station will continue calling these numbers in round robin fashion until the call is answered or until the Conversation Call Timer expires.



**t** TALKAPHONE

Apply Refresh Help Logout

Home  
Maintenance  
Network  
VoIP  
**Number Lists**  
Phone Settings  
Audio Settings  
Call Parameters  
Paging Settings  
Devices  
Digital Outputs  
Voice Messages  
Self Diagnostics & Reporting  
Authentication  
Reboot

### Number Lists

Select a number list to configure

**Enter numbers in selected list :**

1

2

3

4

5

6

**Note**  
To assign a number list to specific buttons, go to Devices > Buttons  
To assign a number list to an auxiliary input, go to Devices > Auxiliary Inputs

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## 7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Talkphone VOIP-500ECK IP Call Station with Avaya IP Office Server Edition.

1. Verify that VOIP-500ECK 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 VOIP-500ECK IP Call Station extension. Verify that the **Current State** is *Idle*.

The screenshot shows the Avaya IP Office System Status application window. The title bar reads "Avaya IP Office System Status - ServerEdition (10.64.110.65) - IP Office Linux PC 11.1.3.0.0 build 23". The main window has a blue header with the Avaya logo and the text "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 (1)
- Extensions (4)
  - 71001
  - 72001 (selected)
  - 72013
  - 72015
- Trunks (4)
- Active Calls
- Resources
- Voicemail
- IP Networking
- Locations

The main content area is titled "Extension Status" and displays the following configuration details for extension 72001:

- Extension Number: 72001
- IP address: 192.168.5.102
- Standard Location: None
- Registrar: Primary
- Telephone Type: Unknown SIP Device
- User-Agent SIP header: ADI VoIP Phone
- Media Stream: RTP
- Layer 4 Protocol: UDP
- Current User Extension Number: 72001
- Current User Name: VOIP-500ECK
- Forwarding: Off
- Twinning: Off
- Do Not Disturb: Off
- Message Waiting: Off
- Phone Manager Type: None
- SIP Device Features: REFER,UPDATE
- License Reserved: No
- Last Date and Time License Allocated: 1/30/2024 1:09:52 PM
- DTMF Required: No
- Packet Loss Fraction: (empty)
- Jitter: (empty)
- Round Trip Delay: (empty)
- Connection Type: (empty)
- Codec: (empty)
- Remote Media Address: (empty)

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	00:38:20			

At the bottom of the main content area, there are several buttons: "Trace", "Trace All", "Pause", "Ping", "Call Details", "Print...", and "Save As...". The bottom status bar shows the time "6:45:22 PM" and the status "Online".

- Alternatively, the Talkphone VOIP-500ECK IP Call Station registration status can be seen on the Web Administration Interface. The SIP Settings screen on the Talkphone VOIP-500ECK IP Call Station shows it registered under **Registration Status**.

The screenshot shows the Talkphone Web Administration Interface. The browser address bar displays the URL `192.168.5.102/Home.aspx#sipsettings.xasp`. The page title is "SIP Settings".

**SIP Settings**

**Assign a phone number :**

Phone Number:

**Specify SIP Server FQDN/IP Address :**

Primary SIP Server FQDN/IP Address:

Secondary SIP Server FQDN/IP Address:

Tertiary SIP Server FQDN/IP Address:

**Enable / disable SIP registration :**

Register

**Specify SIP registrar :**

Username:

Password:

Primary SIP Server IP Address:

Secondary SIP Server IP Address:

Tertiary SIP Server IP Address:

Port:  (Port Range: 1024-49151)

Re-registration Time:  (Range: 10-14400 seconds)

**Specify outbound proxy :**

Username:

Password:

Outbound Proxy 1 IP Address:

Outbound Proxy 2 IP Address:

Outbound Proxy 3 IP Address:

Port:  (Port Range: 1024-49151)

**Registration status :**

Primary registrar is active : Registered as 72001@10.64.110.65: Pri Reg: 0, Sec Reg: 0, Ter Reg: 0

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- Place incoming/outgoing calls to/from Talkphone VOIP-500ECK 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-500ECK 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 using Manager*, Release 11.1, available at <http://support.avaya.com> as an HTML document.
- [2] *Talkphone VOIP-500 Series Phone Configuration and Operation Manual*, v3.0.2, Rev 11/17/22, available at <https://talkphone.com>.



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