Compatible SIP Endpoints

Yeastar S-Series VoIP PBX

Version: 1.0
Updated: December 4, 2019

Support: +86-592-5503301
Support: support@yeastar.com
https://www.yeastar.com

Yeastar Information Technology Co.Ltd.

Contents

Compatible SIP Endpoints

Yeastar have tested multiple IP phones, soft phones, intercom devices, and door phones with Yeastar S-Series VoIP PBX. Refer to the phone registration guides to register your phone to Yeastar S-Series VoIP PBX.

Algo 8180G2 Audio Alert

Algo 8180G2 Test Report

This article is the Interoperability Test Report for Yeastar S-Series VoIP PBX and Algo 8180G2 SIP Audio Alerter.

Tested equipment & software

Equipment	Firmware/Software Version
Algo 8180G2 Audio Alerter	1.7.2
Yeastar S300	30.10.0.59

Summary of test focus

The following table shows a summary of the validated capabilities.

Feature	Test Result
DUT Services	
SIP Registration	PASS
Inbound Call: Ring Extension	PASS
Inbound Call: Page Extension	PASS
Inbound Call: Emergency Alert	PASS
Inbound Call: Multicast	PASS
Serviceability	PASS
PBX Services	
Paging/Intercom Group	PASS

Definitions

Word definitions in the following test plan table.

- **DUT**: Device Under Test, which in this case is the Algo 8180G2 Audio Alerter.
- **Ring Extension**: This is the extension that will be called from Phone A or Phone B in order to trigger a “Ring” sound from the DUT. The DUT will expect to play ring tones, but will not answer the call.
- **Page Extension**: This is the extension that will be called from Phone A or Phone B in order to send paging audio to the DUT. The DUT will answer the call automatically.
- **Announcement Extension**: This is the extension that will be called from Phone A or Phone B in order to play selected announcement in the DUT. The DUT will answers the call automatically.
- **Call to Cancel Extension**: When the DUT is playing the emergency announcement, dial this extension from Phone A or Phone B to cancel the announcement.

- **Phone A**: A SIP compatible endpoint used to call the DUT.
- **Phone B**: A SIP compatible endpoint used to call the DUT and Phone A.
- **Phone C**: A SIP compatible endpoint used to register a Ring/Alert Extension.

Test plan

SIP Registration

The following test cases verify features related to the registration process with Yeastar S300.

Test Case	Expected Result	Test Result
Attempt registering DUT Extension using incorrect password.	Registration failure status is correctly displayed in web interface	PASS
Attempt registering DUT Extension using incorrect username.	Registration failure status is correctly displayed in web interface.	PASS
Correctly register DUT Extension.	DUT registers properly and status is correctly displayed in web interface	PASS
Register DUT multiple extensions.	DUT registers properly and status is correctly displayed in web interface.	PASS
Register DUT Extension using UDP protocol.	DUT registers properly and status is correctly displayed in web interface	PASS
Register DUT Extension using TCP protocol.	DUT registers properly and status is correctly displayed in web interface	PASS
Register DUT Extension using TLS protocol.	DUT registers properly and status is correctly displayed in web interface	PASS

Inbound Call - Ring Extension

The following test cases verify the Ring Extension with different Ring/Alert modes of the DUT.

Test Case	Expected Result	Test Result
Ring/Alert Mode: **Monitor "Ring" event on registered SIP extension**		
Dial Ring Extension from Phone A.	• DUT answers the call automatically and plays the selected ring sound. • DUT continues to ring until the call is canceled by Phone A.	PASS
Ring/Alert Mode: **Use ”Subscribe/Notify” dialog event (RFC4235)** **Note:** Ensure Phone C is registered with the Page Extension.		

Test Case	Expected Result	Test Result
Select **Alert Event** to **Ring**, and call Phone C (Ring Extension registered) from Phone A.	• When Phone C is ringing, DUT plays ring sound. • When Phone C answers the call, DUT stops playing ring sound.	PASS
Select **Alert Event** to **In-Use**, and call Phone C (Ring Extension registered) from Phone A.	• When Phone C is ringing, DUT doesn't play ring sound. • When Phone C answers the call, DUT starts playing ring sound. • When Phone C ends the call, DUT stops playing ring sound.	PASS
Select **Alert Event** to **Ring&In-Use**, and call Phone C (Ring Extension registered) from Phone A.	• When Phone C is ringing, DUT plays ring sound. • When Phone C answers the call, DUT replays the ring sound. • When Phone C ends the call, DUT stops playing ring sound.	PASS
Ring/Alert Mode: **Use "Subscribe/Notify" presence event (RFC 3856/3863 PIDF)**		Not Supported

Inbound Call - Page Extension

The following test cases verify the inbound paging feature of the DUT.

Test Case	Expected Result	Test Result
Dial Page Extension from Phone A.	• DUT answers and a one-way audio page is established from Phone A to UUT. • The call is terminated by hanging up Phone A.	PASS
Dial Page Extension from Phone A and mute/unmute the call.	• Mute: The DUT doesn't plays the audio from Phone A. • Unmute: The DUT plays the audio from Phone A.	PASS
When the Page Extension. is already in a call with Phone A, dial the Page Extension from Phone B.	• Phone B receives busy tone (DUT configured to allow only one simultaneous Page call).	PASS
Dial Page Extension from Phone A and maintain the call for a period of time.	• The call remains up after the Session Refresh (REINVITE) is sent to the DUT.	PASS

Inbound Call - Emergency Alert

The following test cases verify the inbound Emergency Alert feature of the DUT.

Test Case	Expected Result	Test Result
Dial **Announcement** Extension from Phone A.	• DUT answers the call automatically and plays the selected announcement. • DUT keeps playing the selected announcement even the call is canceled by Phone A.	PASS
When DUT is playing an announcement, dial **Call to Cancel** Extension from Phone A.	• DUT stops playing the selected announcement.	PASS

Inbound Call: Multicast

The following test cases verify the Multicast Master/Sender feature on the DUT. The DUT acts as a multicast master.

Test Case	Expected Result	Test Result
Prerequisite: • On the DUT, set the Multicast mode to Master/Sender and configure the multicast IP address and port. • On the DUT, register a Zone 1 Page Extension. • On the other phones, configure the same multicast IP address and port as the DUT to receive multicast.		
Dial the Zone 1 Page Extension from Phone A.	• DUT plays the selected Page Tone and plays the audio from Phone A. • The other phones plays the DUT selected Page Tone and plays the audio from Phone A.	PASS

PBX Feature: Paging/Intercom Group

The following test cases verify the Paging/Intercom Group of Yeastar S300. The DUT acts as a multicast slaver.

Test Case	Expected Result	Test Result
Verify PBX feature: 1-Way Multicast Paging. **Prerequisite**: • On Yeastar S300, add a 1-Way Multicast Paging group. • On the DUT, set the Multicast mode to Slave/Receiver and configure the same multicast IP address and port as the Yeastar S300.		
Dial the 1-Way Multicast Paging number from Phone A.	DUT answers the call automatically, and the 1-way paging is established.	PASS

Test Case	Expected Result	Test Result
Cancel the call by hanging up Phone A.	DUT ends the call and stops playing the paging audio.	PASS
Verify PBX feature: 1-Way Paging. **Prerequisite**: • On Yeastar S300, add a 1-Way Paging group. • On the DUT, register a Page Extension. The page extension is a member of the 1-Way paging group.		
Dial the 1-Way Paging number from Phone A.	DUT answers the call automatically, and the 1-way paging is established.	PASS
Cancel the call by hanging up Phone A.	DUT ends the call and stops playing the paging audio.	PASS
Verify PBX feature: 2-Way Intercom. **Prerequisite**: • On Yeastar S300, add a 2-Way Intercom group. • On the DUT, register a Page Extension. The page extension is a member of the 2-Way Intercom group.		
Dial the 2-Way Intercom number from Phone A.	DUT answers the call automatically, and the 2-way intercom is established.	PASS
Cancel the call by hanging up Phone A.	DUT ends the call and stops playing the audio.	PASS

Serviceability

The following test cases verify the serviceability of the DUT.

Test Case	Expected Result	Test Result
Disconnect, then reconnect, the ethernet cable from the DUT.	DUT registers with the PBX server after the network is restored.	PASS

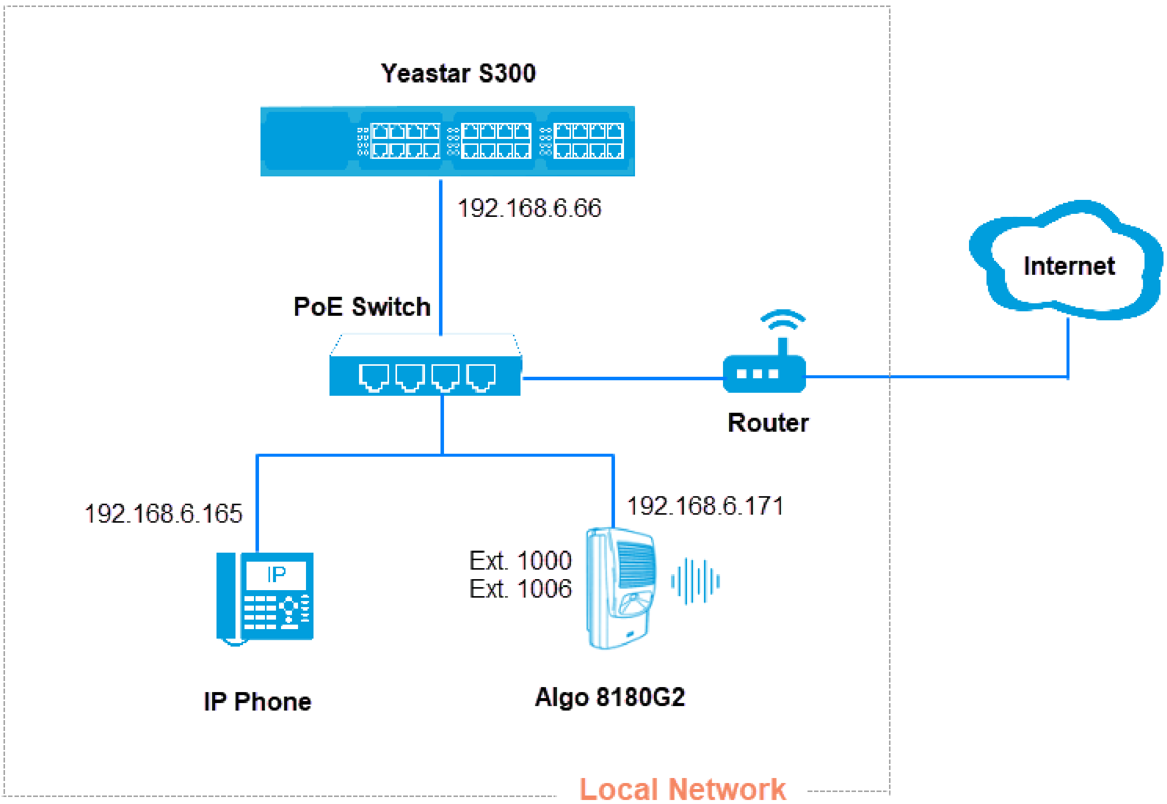
Register Algo 8180G2 Audio Alerter with Yeastar S-Series VoIP PBX

This guide describes the configuration steps required for Algo 8180G2 SIP Audio Alerter to interoperate with Yeastar S-Series VoIP PBX.

Below is a guideline of how to register a Ring extension and a Page extension on Algo 8180G2.You may need to configure the other settings of the Algo 8180G2 Audio Alerter depending on your VoIP solution.

Network Topology

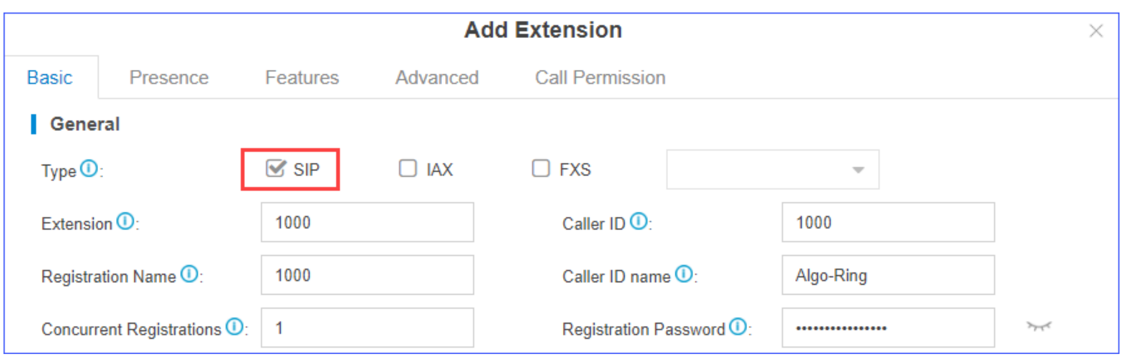
The following diagram shows how the testing network is configured for reference.



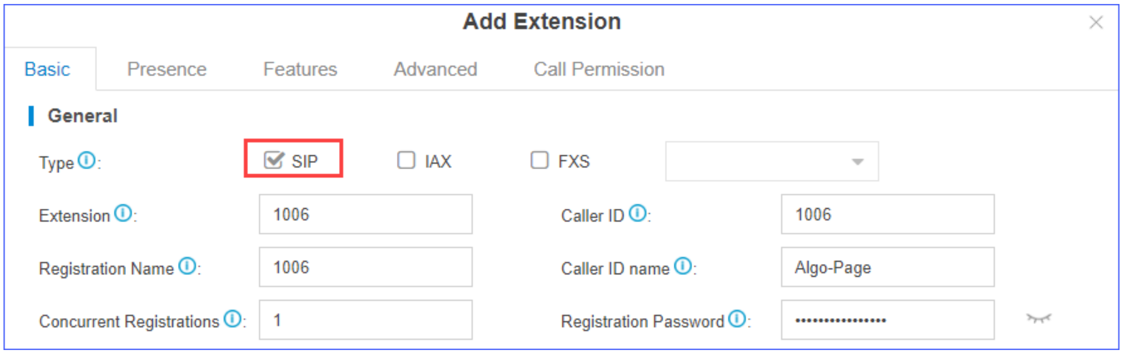
Yeastar S300 configuration

Add two SIP extensions on Yeastar S300, and provide the extension details in Algo 8101G2 web page.

1. Log in Yeastar S300 web interface, go to **Settings→PBX→Extensions**.
2. Add an extension, this extension will be registered as the Algo Ring extension.
    a. Click **Add**.
    b. Leave the default settings or change the General settings according to your needs.
    c. Click **Save** and **Apply**.



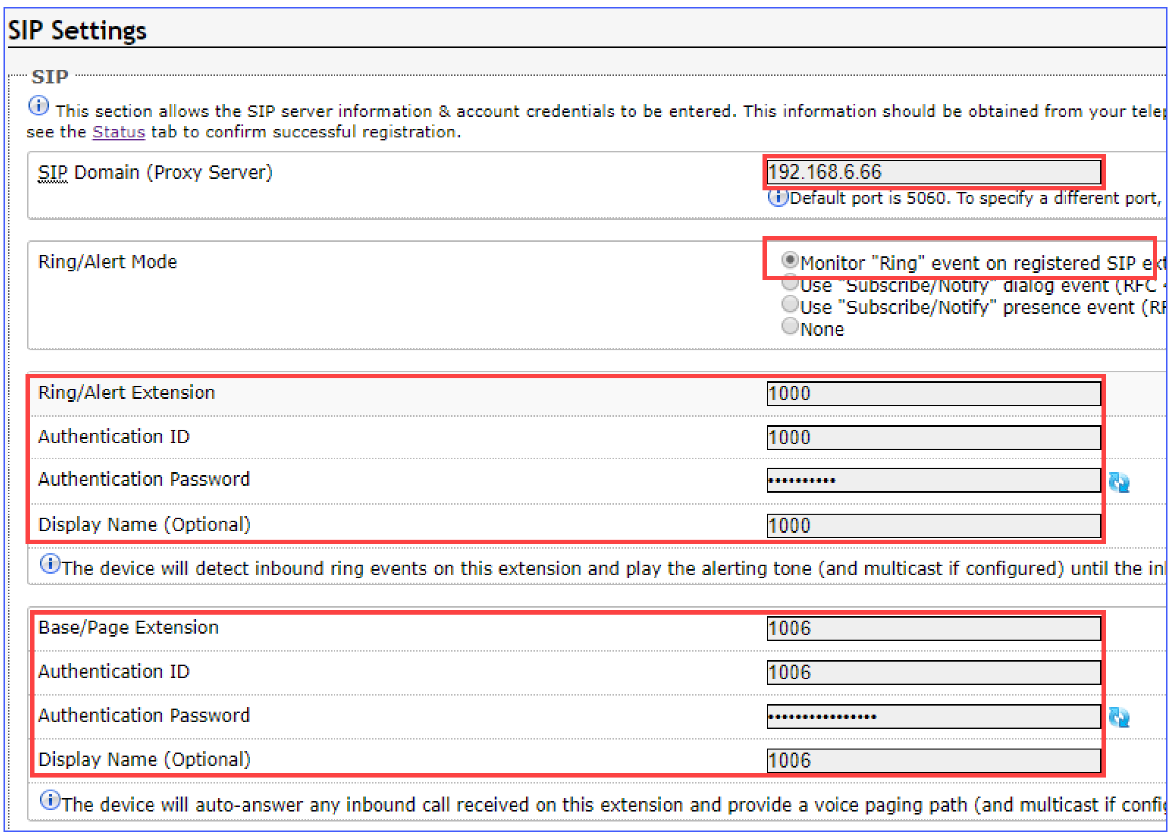
3. Add an extension, this extension will be registered as the Algo Page extension.
    a. Click **Add**.
    b. Leave the default settings or change the General settings according to your needs.
    c. Click **Save** and **Apply**.



Algo 8180G2 configuration

1. Access the Algo 8180G2 web interface, enter the password, and click **Login**.

   The default password is *algo*.
2. Go to **Basic Settings**→**SIP**, enter the following settings.



- **SIP Domain (Proxy Server)**: Enter the IP address of Yeastar S-Series VoIP PBX.
- **Ring/Alert Mode**: Select **Monitor "Ring" event on registered SIP extension**.
- **Ring/Alert Extension**

  Enter the extension details of Ring/Alert extension.

  - **Ring/Alert Extension**: Enter the extension number.
  - **Authentication ID**: Enter the extension's **Registration Name**.
  - **Authentication Password**: Enter the extension's **Registration Password**.

- **Base/Page Extension**

  Enter the extension details of Base/Page extension.

  - **Ring/Alert Extension**: Enter the extension number.
  - **Authentication ID**: Enter the extension's **Registration Name**.
  - **Authentication Password**: Enter the extension's **Registration Password**.

3. Click **Save**.
4. Go to **Status** to check the registration status.

   If the extension is registered successfully, the status will display "Successful".

Result:

- When you dial the Ring/Alert extension 1000, the Algo 8180G will play ring tones until the you hang up the call.

  Note: The call is not answered.

- When you dial the Base/Page extension 1006, the Algo 8180G will answer the call automatically.

Algo 8201 SIP Intercom

Algo 8201 SIP Intercom Test Report

This article is the Interoperability Test Report for Yeastar S-Series VoIP PBX and Algo 8201 SIP Intercom.

Tested Equipment & Software

Equipment	Firmware/Software Version
Algo 8201 SIP Intercom	1.6.2
Yeastar S300	30.10.0.59

Summary of test focus

The following table shows a summary of the validated capabilities.

Feature	Test Result
DUT Services	
SIP Registration	PASS
Inbound Call	PASS
Outbound Call	PASS
Serviceability	PASS
PBX Services	

Feature	Test Result
Paging/Intercom Group	PASS

Definitions

Word definitions in the following test plan table.

- **DUT**: Device Under Test, which in this case is the Algo 8180G2 Audio Alerter.
- **Phone A**: A SIP compatible endpoint used to place and receive calls.
- **Phone B**: A SIP compatible endpoint used to place and receive calls.

Test plan

SIP Registration

Test Case	Expected Result	Test Result
Attempt registering DUT Extension using incorrect password.	Registration failure status is correctly displayed in web interface	PASS
Attempt registering DUT Extension using incorrect username.	Registration failure status is correctly displayed in web interface.	PASS
Correctly register DUT Extension	DUT registers properly and status is correctly displayed in web interface	PASS
Register DUT Extension using UDP protocol.	DUT registers properly and status is correctly displayed in web interface	PASS
Register DUT Extension using TCP protocol.	DUT registers properly and status is correctly displayed in web interface	PASS
Register DUT Extension using TLS protocol.	DUT registers properly and status is correctly displayed in web interface	PASS

Inbound Call

Test Case	Expected Result	Test Result
Call the DUT from Phone A.	A two-way audio call is established.	PASS
Call the DUT from Phone A and mute/un-mute the call.	• Mute: The DUT doesn't plays the audio from Phone A. • Unmute: The DUT plays the audio from Phone A.	PASS
When the DUT is already in a call with Phone A, call the DUT from Phone B.	Phone B receives busy tone while Phone A call continues.	PASS
Call the DUT from Phone A and maintain the call for a period of time.	The call remains up after the Session Refresh (REINVITE) is sent to the DUT.	PASS

Outbound Call

Test Case	Expected Result	Test Result
Press the call button on the DUT to call Phone A.	When the call is answered by Phone A. a two-way audio call is established.	PASS
Call the DUT from Phone A and Phone A doesn't answer the call.	Phone A continues ringing until timeout.	PASS
When an outbound call is established on the DUT and Phone A, call the DUT from Phone B.	Phone B receives busy tone, while Phone A call continues.	PASS

Serviceability

Test Case	Expected Result	Test Result
Disconnect, then reconnect, the ethernet cable from the DUT.	DUT registers with the PBX server after the network is restored.	PASS

PBX Feature: Paging/Intercom Group

The following test cases verify the Paging/Intercom Group of Yeastar S300. The DUT acts as a multicast slaver.

<table>
<tr><th>Test Case</th><th>Expected Result</th><th>Test Result</th></tr>
<tr><td colspan="3">Verify PBX feature: 1-Way Multicast Paging.
Prerequisite:
On Yeastar S300, add a 1-Way Multicast Paging group.
On the DUT, set the Multicast mode to Slave/Receiver and configure the same multicast IP address and port as the Yeastar S300.</td></tr>
<tr><td>Dial the 1-Way Multicast Paging number from Phone A.</td><td>DUT answers the call automatically, and the 1-way paging is established.</td><td>PASS</td></tr>
<tr><td>Cancel the call by hanging up Phone A.</td><td>DUT ends the call and stops playing the paging audio.</td><td>PASS</td></tr>
<tr><td colspan="3">Verify PBX feature: 1-Way Paging.
Prerequisite:
On Yeastar S300, add a 1-Way Paging group.
On the DUT, register a Page Extension.
The page extension is a member of the 1-Way paging group.</td></tr>
<tr><td>Dial the 1-Way Paging number from Phone A.</td><td>DUT answers the call automatically, and the 1-way paging is established.</td><td>PASS</td></tr>
<tr><td>Cancel the call by hanging up Phone A.</td><td>DUT ends the call and stops playing the paging audio.</td><td>PASS</td></tr>
</table>

Test Case	Expected Result	Test Result
Verify PBX feature: 2-Way Intercom. **Prerequisite**: • On Yeastar S300, add a 2-Way Intercom group. • On the DUT, register a Page Extension. The page extension is a member of the 2-Way Intercom group.		
Dial the 2-Way Intercom number from Phone A.	DUT answers the call automatically, and the 2-way intercom is established.	PASS
Cancel the call by hanging up Phone A.	DUT ends the call and stops playing the audio.	PASS

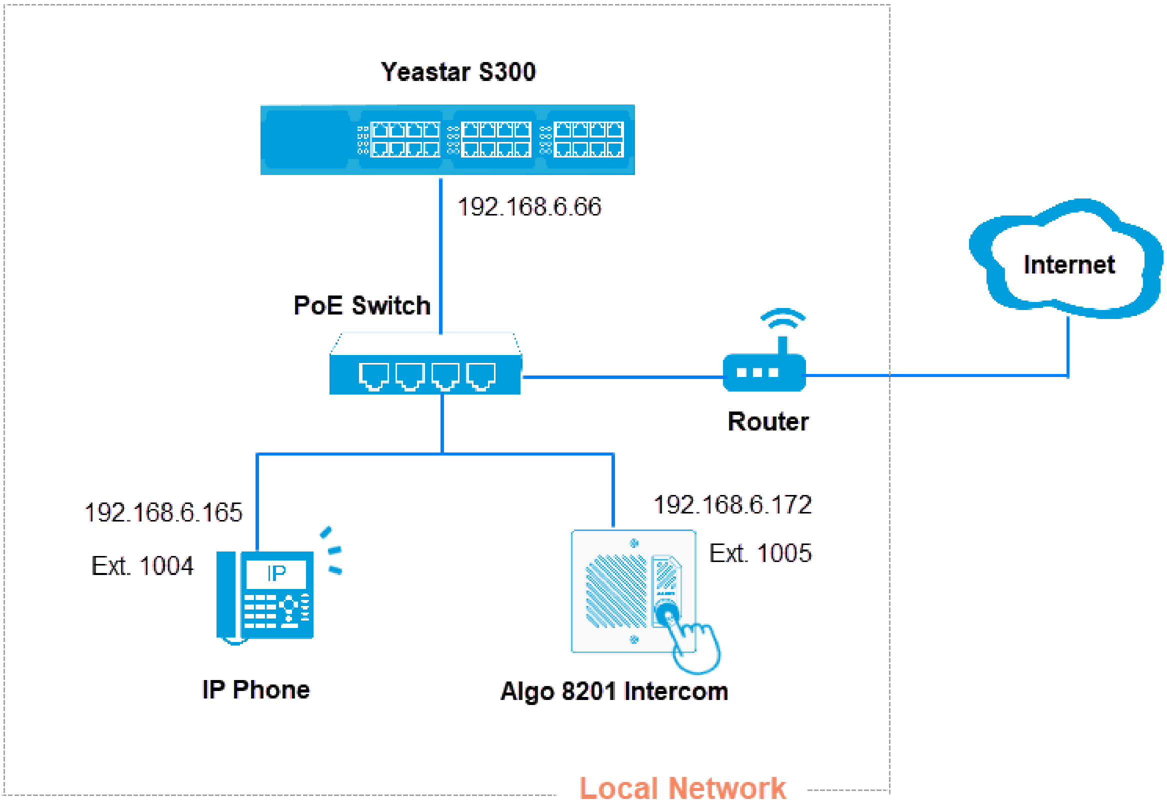
Register Algo 8201 SIP Intercom with Yeastar S-Series VoIP PBX

This guide describes the configuration steps required for Algo 8201 SIP Intercom to interoperate with Yeastar S-Series VoIP PBX.

Below is a guideline of how to register an extension on Algo 8201 SIP Intercom. You may need to configure the other settings of the Algo 8201 SIP Intercom depending on your VoIP solution.

Network Topology

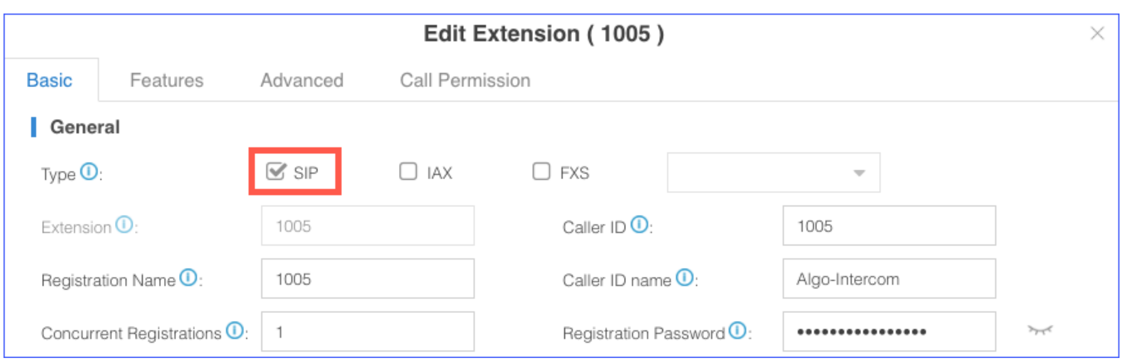
The following diagram shows how the testing network is configured for reference.



Yeastar S300 configuration

Add a SIP extension on Yeastar S300, and provide the extension details in Algo 8201 web page.

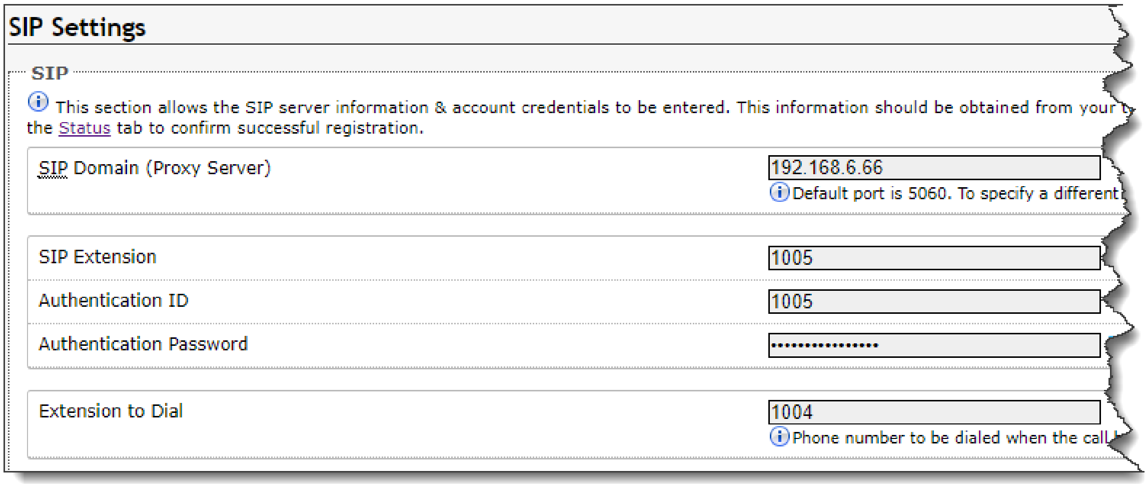
1. Log in Yeastar S300 web interface, go to **Settings**→**PBX**→**Extensions**.
2. Add an extension, this extension will be registered as the Algo Ring extension.
    a. Click **Add**.
    b. Leave the default settings or change the General settings according to your needs.
    c. Click **Save** and **Apply**.



Algo 8201 SIP Intercom configuration

1. Access the Algo 8201 web interface, enter the password, and click **Login**.

   The default password is *algo*.
2. Go to **Basic Settings**, enter the following settings:



- **SIP Domain (Proxy Server)**: Enter the IP address of Yeastar S-Series VoIP PBX.
- **SIP Extension**: Enter the extension number.
- **Authentication ID**: Enter the extension's **Registration Name**.
- **Authentication Password**: Enter the extension's **Registration Password**.
- **Extension to Dial**: Enter an extension of Yeastar S-Series VoIP PBX. When a visitor presses the blue call button on Algo 8201, the extension will be dialed.

3. Click **Save**.
4. Go to **Status** to check the registration status.

   If the extension is registered successfully, the status will display "Successful".

Result:

- When a visitor presses the blue call button on the Algo 8201 SIP Intercom, the extension 1004 will ring.

ALCATEL Phone

Register ALCATEL Phone with Yeastar S-Series VoIP PBX

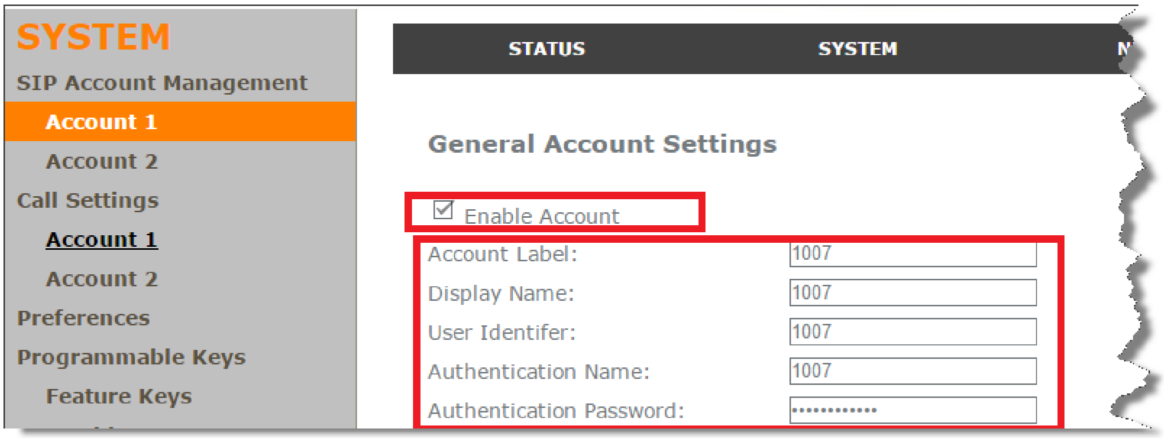
This article is based on ALCATEL Temporis IP151 v1. 1. 0. B and Yeastar S-Series VoIP PBX v30.8.0.14.

Configure the IP address via phone user interface

1. Press **System→Network→Basic Settings→Dual Mode→WAN Setting**.
2. Choose **Static IP** and alter the **IP Address, Subnet Mask**, **Preferred DNS Server**, **Alternate DNS Server**.
3. **Apply** it after inputting the correct information.
4. **Reboot** the phone and log in the phone web user interface using the new IP address.
5. Enter the user name and password, click **Log In** to enter the web user interface.
   - User Name: admin
   - Default Password: admin

Account Registration

1. Log in the IP phone, go to **System→SIP Account Management**, select one account to configure.
2. Enable the account and fill in the extension information.



- **Enable Register**: check
- **Account Label**: The name you want to display on the phone screen.
- **Display Name**: The name you want to display on another person's phone screen when you are calling the phone.
- **User Identifier**: Enter the extension's **Caller ID**.
- **Authentication Name**: Enter the extension's **Registration Name**.
- **Authentication Password**: Enter the extension's **Registration Password**.

3. In the **SIP Server** section and **Registration** section, fill in your PBX information.

- **SIP Server**
  - **Server Address:** Enter the domain or IP address of your PBX.
  - **Server Port:** Enter the SIP port of your PBX.
- **Registration**
  - **Server Address:** Enter the domain or IP address of your PBX.
  - **Port:** Enter the SIP port of your PBX.

4. Click **Apply**.

If the registration is successfully, the register status would show "Registered".

Cisco

Register Cisco Phone with Yeastar S-Series VoIP PBX

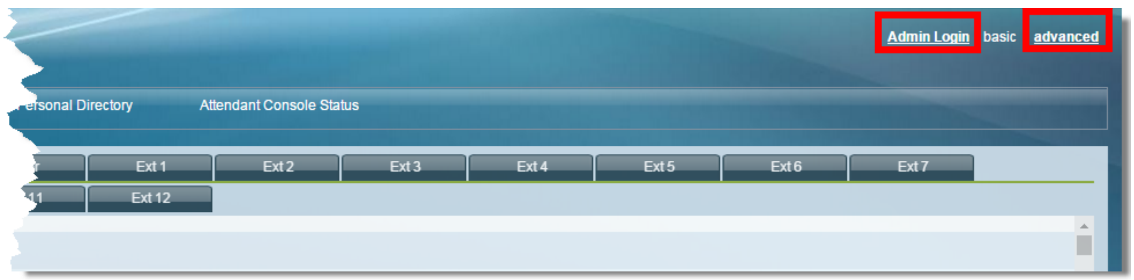
This article is based on Cisco SPA509G and Yeastar S-Series VoIP PBX v30.8.0.14.

This guide is applicable to the following phones:

- Cisco SPA series: 301, 303, 501G, 502G, 508G, 509G, 512G, 514G, 525G5
- Cisco CP7821

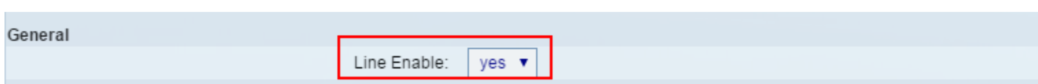
Note: For the IP phone with different firmware version, the web GUI may be different.

1. To check the IP address of the phone, press the menu key, go to **Network**, then press **Select**.
2. Type the phone IP address in your browser, click **Enter** key to access the web page of the IP phone.
3. In the upper-right corner, click **Admin Login**, then click **Advanced** to access the advanced administrator page.

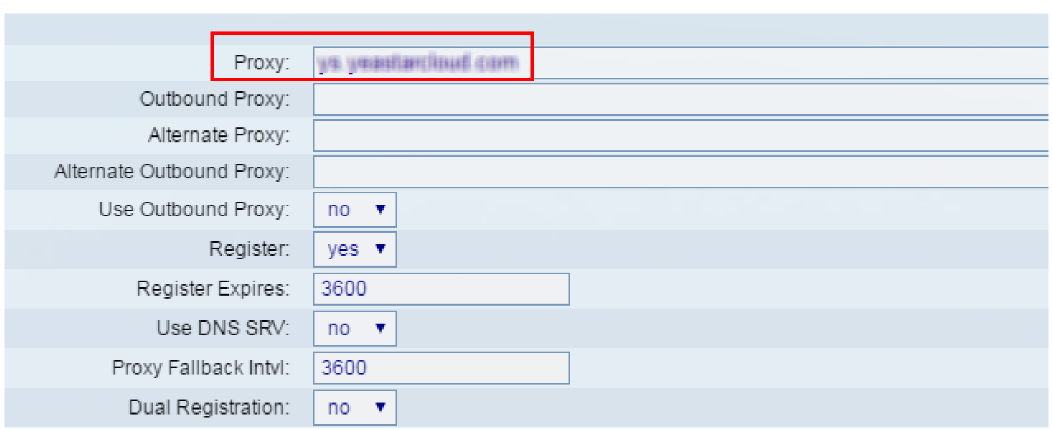


   - Choose one account to configure. Here we click **EXT1** to configure account 1. Configure the account as follows:

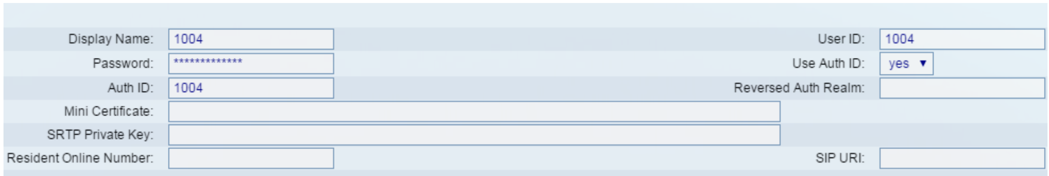
4. Choose one account to configure. Here we choose **EXT1**.
   a. Set the **Line Enable** to **Yes**.



   b. In the **Proxy and Registration** section, set the **Proxy** to the domain or IP address of your PBX.

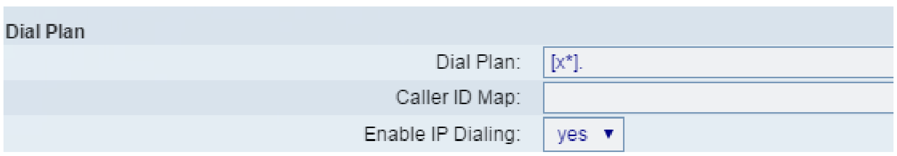


   c. In the **Subscriber Information** section, fill in the extension information.

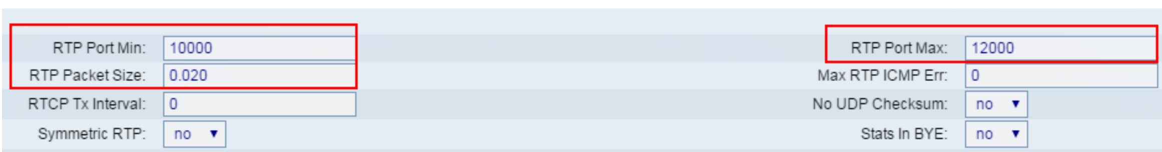


- **Display Name**: Set the name you want to appear on other phone's display when calling other phones.
- **User ID**: Fill in the extension number.
- **Password**: Fill in the extension's **Registration Password**.
- **Use Auth ID**: Set to **Yes**.
- **Auth ID**: Fill in the extension's **Registration Name**.

d. In the **Dial Plan** section, set the **Dial Plan** to `[x*].`.



5. Click **Phone** tab, adjust the audio parameters according to the RTP settings on your PBX, and set **RTP Packet Size** to **0.020**.



6. In the bottom of the page, click **Submit All Changes**.

The phone will restart. After the phone restarts, check if the extension is registered.

Fanvil

Register Fanvil Phone with Yeastar S-Series VoIP PBX

This guide is based on Fanvil C400 and Yeastar S-Series VoIP PBX v30.8.0.14.

This guide is applicable to the following phones:

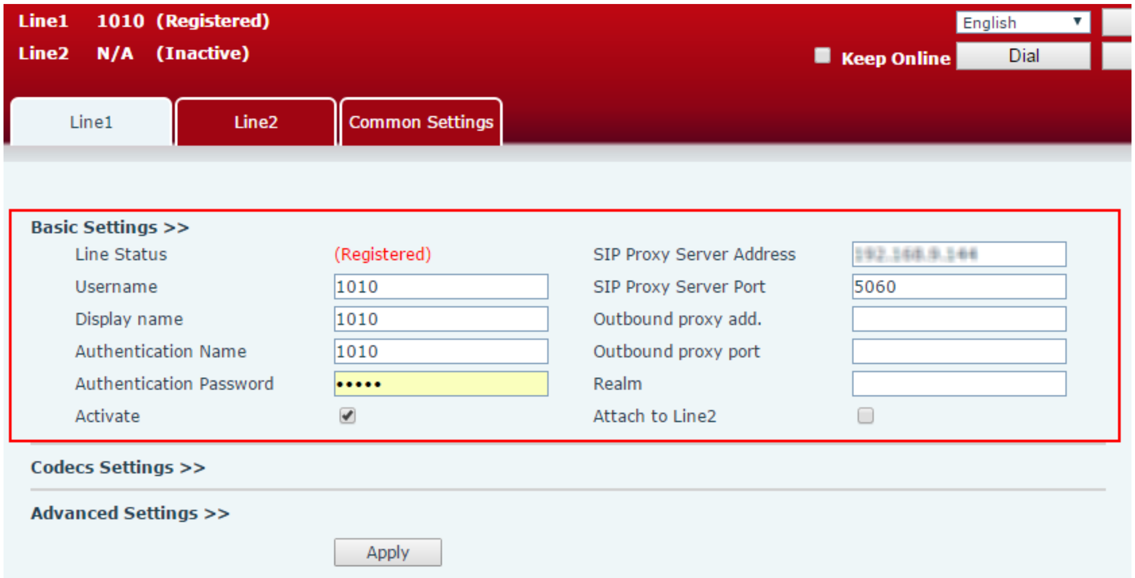
- Fanvil C Series: C01, C58, C58P, C400, C600
- Fanvil X3 Series: X3, X3P, X3SP
- Fanvil X5 Series: X5,X5G

Note: For the IP phone with different firmware version, the web GUI may be different.

1. Log in the web page of the phone.

- **User**: admin
- **Password**: admin

2. Click **Line** and choose a line to configure.



- **User Name**: Fill in the extension number.
- **Display Name**: Set the name you want to appear on other phone's screen when calling other phones.
- **Authentication Name**: Fill in the extension's **Registration Name**.
- **Authentication Password**: Fill in the extension's **Registration Password**.
- **Active**: Check
- **SIP Proxy Server Address**: Fill in the domain or IP address of your PBX.
- **SIP Proxy Server Port**: Fill in the SIP port of your PBX.

3. Click **Apply**.

   If the extension is registered, the **Line Status** will show "Registered".

Grandstream

Register Grandstream Phone with Yeastar S-Series VoIP PBX

This guide is based on Grandstream GXP2135 and Yeastar S-Series VoIP PBX v30.8.0.14.

This guide is applicable to the following phones:

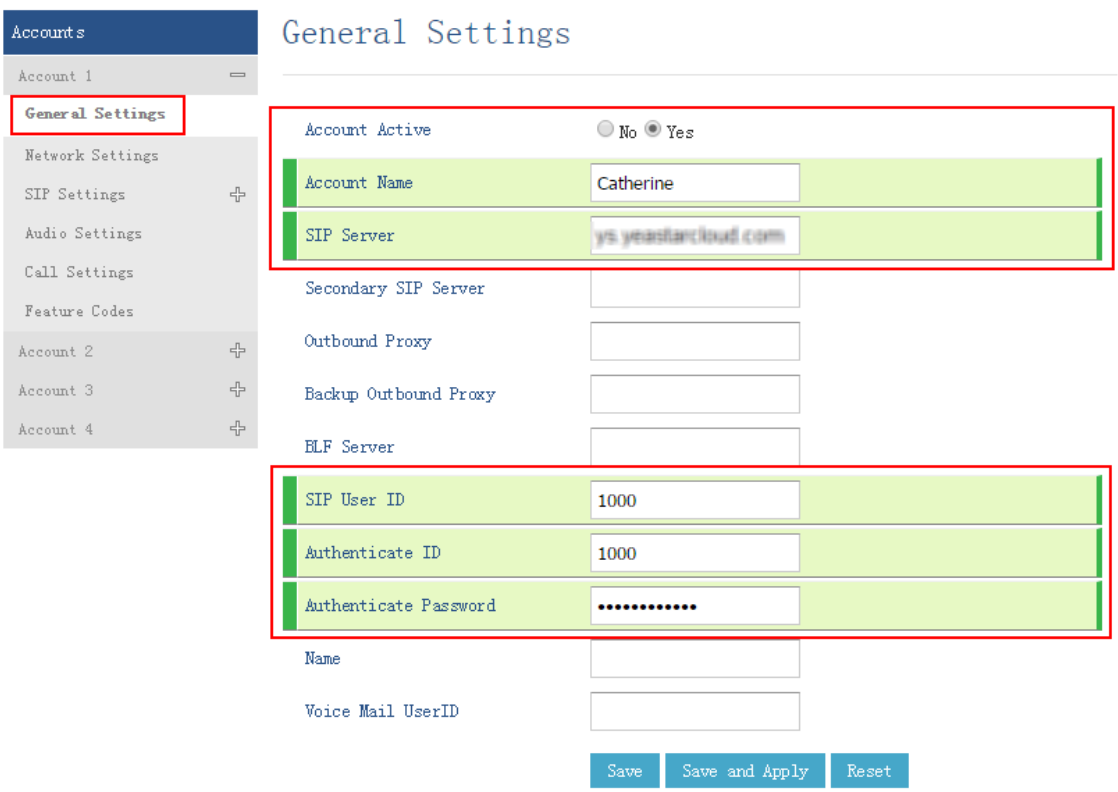
- Grandstream GXP Series 1160, 1165, 1400, 1405, 1450, 1610, 1620, 1625, 1628, 1630, 2130, 2135, 2140, 2160, 2170, 2200, 3240, 3245

Note: For the IP phone with different firmware version, the web GUI may be different.

1. Log in the web page of the IP phone.

- **Username**: admin
- **Default Password**: admin

2. Click **Account** tab, choose one account, and configure the general settings.



- **Account Active**: Yes
- **Account Name**: Set a name for the account, the name will be displayed on the phone LCD.
- **SIP Server**: Fill in the domain or IP address of your PBX.
- **SIP User ID**: Fill in the extension number
- **Authenticate ID**: Fill in the extension's **Registration Name**.

- **Authenticate Password**: Fill in the extension's **Registration Password**.

3. Click **Save and Apply**.
4. Go to **Status**→**Account Status** to check the account status.

   If the extension is registered, the **SIP Registration** shows "Yes".

Hikvision

Test Report for Hikvision DS-KH6320 Video Intercom Indoor Station

This article is the Interoperability Test Report for Yeastar S-Series VoIP PBX and Hikvision DS-KH6320 Video Intercom Indoor Station.

Tested equipment & software

Equipment	Firmware/Software Version
Hikvision DS-KH6320 Video Intercom Indoor Station	2.0.2
Yeastar S100	30.10.0.67

Summary of test focus

The following table shows a summary of the validated capabilities.

Feature	Test Result
DUT Services	
SIP Registration	PASS
Inbound Call (audio)	PASS
Outbound Call (audio)	PASS
Inbound Call (video)	PASS
Outbound Call (video)	FAIL

Feature	Test Result
Serviceability	PASS

Definitions

Word definitions in the following test plan table.

- **DUT**: Device Under Test, which in this case is the DS-KH6320 Video Intercom Door Station.
- **Phone A**: A SIP compatible endpoint used to call the DUT.
- **Phone B**: A SIP compatible endpoint registered on DS-KD8003, used to call the DUT.
- **Phone C**: A SIP compatible endpoint registered on Yealink T58A Video Phone, used to call the DUT for video call.

Test plan

SIP Registration

The following test cases verify features related to the registration process with Yeastar S100.

Test Case	Expected Result	Test Result
Attempt registering DUT Extension using incorrect password.	Registration failure status is correctly displayed in web interface.	PASS
Attempt registering DUT Extension using incorrect username.	Registration failure status is correctly displayed in web interface.	PASS
Correctly register DUT Extension.	DUT registers properly and status is correctly displayed in web interface.	PASS
Register DUT multiple extensions.	DUT registers properly and status is correctly displayed in web interface.	PASS
Register DUT Extension using UDP protocol.	DUT registers properly and status is correctly displayed in web interface.	PASS

Inbound Call

The following test cases verify the inbound calling capability of the DUT.

Test Case	Expected Result	Test Result
Audio call		

Test Case	Expected Result	Test Result
Call DUT from Phone A.	• DUT answers the call and verify that a two-way audio is established. • DUT continues to ring until the call is canceled by Phone A.	PASS
Video call		
Call DUT from Phone B.	• Verify that a two-way audio call is established. • When DUT answered the call, DUT can monitor the outdoor station in real-time.	PASS

Outbound call

The following test cases verify the inbound calling capability of the DUT.

Test Case	Expected Result	Test Result
Audio Call		
Dial extension number on the DUT to call Phone A.	When the call is answered by Phone A, a two-way audio call is established.	PASS
Call the DUT from Phone A and Phone A doesn't answer the call.	Phone A continues ringing until timeout.	PASS
When an outbound call is established on the DUT and Phone A, call the DUT from Phone B.	Phone B receives busy tone while Phone A call continues.	PASS
Video Call		
Dial extension number on the DUT to call Phone C.	Phone C answered the call, both DUT and phone C can get video call from each other.	FAIL

Serviceability

Test Case	Expected Result	Test Result
Disconnect, then reconnect the Ethernet cable from the DUT.	DUT registers with the PBX server after the network is restored.	PASS

Test Report for Hikvision DS-KD8003 Video Intercom Door Station

This article is the Interoperability Test Report for Yeastar S-Series VoIP PBX and Hikvision DS-KD8003 Video Intercom Door Station.

Tested equipment & software

Equipment	Firmware/Software Version
Hikvision DS-KD8003 Video Intercom Door Station	2.1.0
Yeastar S100	30.10.0.67

Summary of test focus

The following table shows a summary of the validated capabilities.

Feature	Test Result
DUT Services	
SIP Registration	PASS
Inbound Call (audio)	FAIL
Outbound Call (audio)	PASS
Inbound call (video)	PASS
Outbound call (video)	PASS
Serviceability	PASS

Definitions

Word definitions in the following test plan table.

- **DUT**: Device Under Test, which in this case is the DS-KD8003 Video Intercom Door Station.
- **Phone A**: A SIP compatible endpoint used to call the DUT.
- **Phone B**: A SIP compatible endpoint used to call the DUT and Phone A.
- **Phone C**: A SIP compatible endpoint registered on Yealink T58A video phone for video call.

Test plan

SIP Registration

The following test cases verify features related to the registration process with Yeastar S100.

Test Case	Expected Result	Test Result
Attempt registering DUT Extension using incorrect password.	Registration failure status is correctly displayed in web interface.	PASS
Attempt registering DUT Extension using incorrect username.	Registration failure status is correctly displayed in web interface.	PASS
Correctly register DUT Extension.	DUT registers properly and status is correctly displayed in web interface.	PASS
Register DUT multiple extensions.	DUT registers properly and status is correctly displayed in web interface.	PASS
Register DUT Extension using UDP protocol.	DUT registers properly and status is correctly displayed in web interface.	PASS

Inbound Call

The following test cases verify the inbound calling capability of the DUT.

Test Case	Expected Result	Test Result
Audio call		
Call DUT from Phone A.	DUT will be able to answer the call and phone A can get the audio from DUT site.	FAIL
Video call		
Call DUT from Phone C.	DUT will auto answer the call, video call will be established with DUT. Phone C can monitor DUT status and DUT can not see the status on Phone C.	PASS

Outbound call

The following test cases verify the inbound calling capability of the DUT.

Test Case	Expected Result	Test Result
Audio Call		

Test Case	Expected Result	Test Result
Press the call button on the DUT to call Phone A.	When the call is answered by Phone A, a two-way audio call is established.	PASS
Call the DUT from Phone A and Phone A doesn't answer the call.	Phone A continues ringing until timeout.	PASS
When an outbound call is established on the DUT and Phone A, call the DUT from Phone B.	Phone B receives busy tone for leaving message.	PASS
Video Call		
Press the call button on the DUT to call Phone C.	When Phone C answered the call, Phone C can monitor the outdoor status in real-time, but DUT fails to monitor the status on Phone C side.	PASS

Serviceability

The following test cases verify the serviceability of the DUT.

Test Case	Expected Result	Test Result
Disconnect, then reconnect the Ethernet cable from the DUT.	DUT registers with the PBX server after the network is restored.	PASS

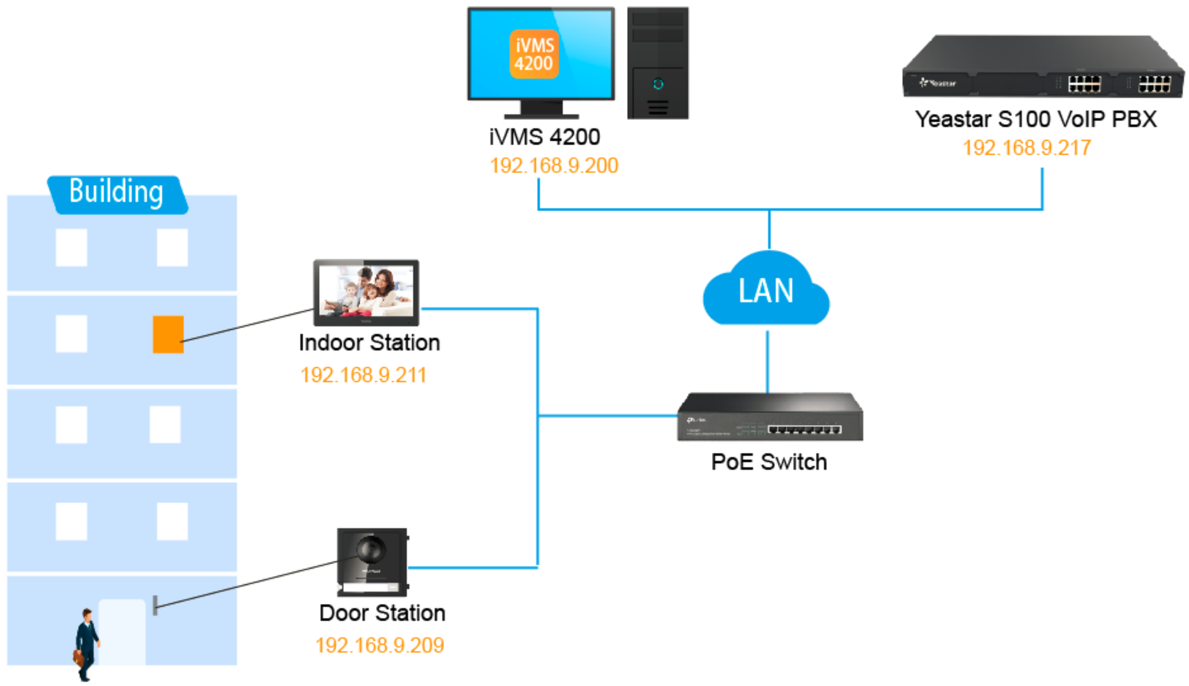
Integrate Yeastar S-Series VoIP PBX with Hikvision Intercom Video Devices

By integrating Yeastar S-Series VoIP PBX with Hikvision Video Intercom Indoor & Door Station, you can establish video & audio call between the Indoor Station and the Door Station.

Local Network (Tested Environment)

In this guide, the Hikvision devices and Yeastar PBX are in the same local network.

Figure 1: Local Network Topology



The following table shows the information of the tested environment.

Device	Firmware Version	IP Address
Yeastar S100	30.10.0.67	192.168.9.217
Hikvision DS-KH6320 Indoor Station	V2.0.2	192.168.9.211
Hikvision DS-KD8003 Door Station	V2.1.0	192.168.9.209

Public Network

If your PBX is not in the same network with Hikvision devices, you need to do port forwarding on the router.

- SIP registration port: Default UDP 5060
- RTP ports: Default UDP 10000-12000

Note: Hikvision devices and the iVMS 4200 client must in the same local network.

Preparation: Configure Hikvision iVMS-4200 Client

iVMS-4200 Client Software is a management software for Hikvision Devices. You can manage devices on the client, including adding, modifying and deleting devices. You can also perform operations such as checking online users and QR code for devices.

Install Hikvision iVMS-4200

Download and install iVMS-4200 Client Software on your local PC.

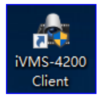
Register and Log in Hikvision iVMS-4200

You should register a super user and then you can log in the client with the super user account as administrator.

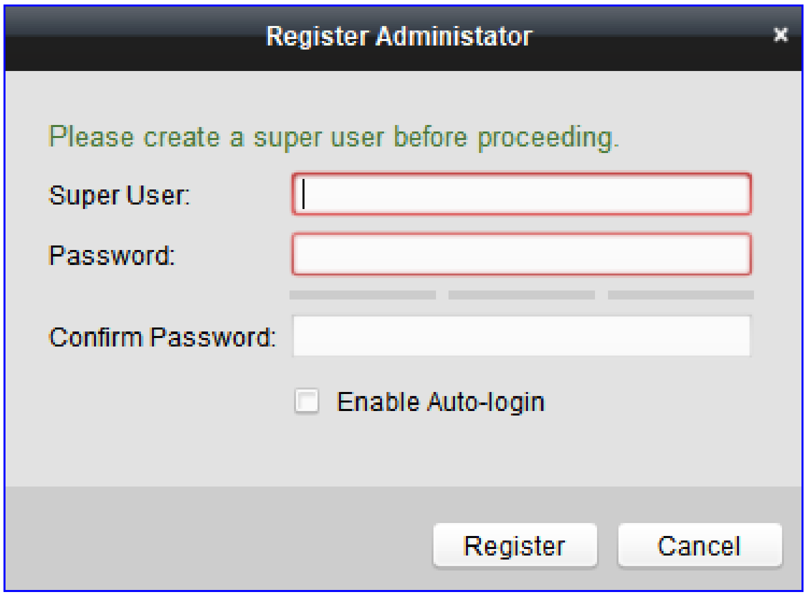
- **Register a User**

  For the first time using the client software, you should register a super user for login.

  Perform the following steps to register a super user for login.

  1. After installing the client, double click 

to run the software.

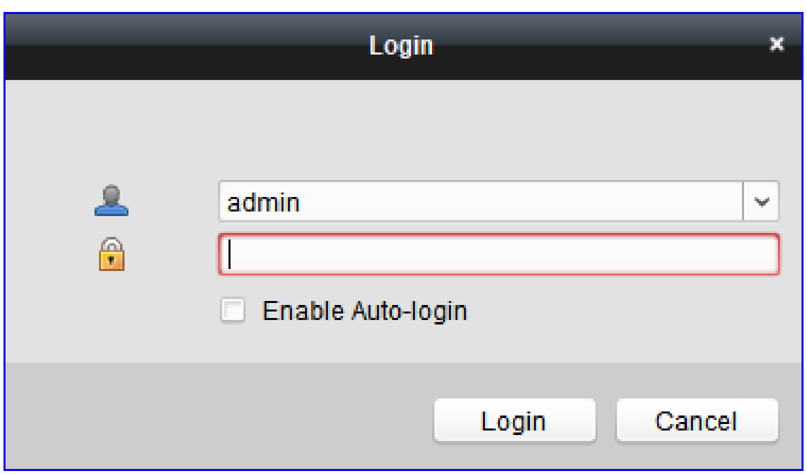
     

  2. Create a user name and password for the super user.
  3. Confirm the password.
  4. **Optional**: Check the **Enable Auto-login** checkbox to log in to the software automatically.
  5. Click **Register** to register the super user.

- **Login**

  Perform this task if you want to log in to the client software.

  1. Run the client software to open login dialog.

  2. Input the user name and password you registered.
  3. Check the **Enable Auto-login** checkbox to log in to the software automatically for next running.

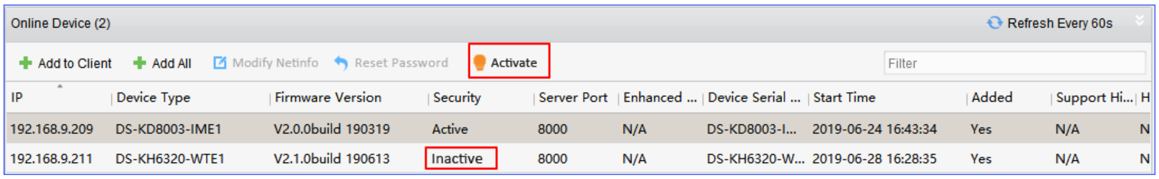
4. Click **Login** to log in to the client software.

Activate Intercom Video Devices

For some devices, you are required to create the password to activate them before they can be added to the software and work properly.

Perform this task to activate device:

1. Go to **Device Management→ Online Device**.
2. Check the device status (shown on **Security** column) and select an inactive device.



3. Click **Activate** to open the **Activation** dialog.
4. Create a password in the **Password** field and confirm the password.

Add Devices to iVMS-4200 Client

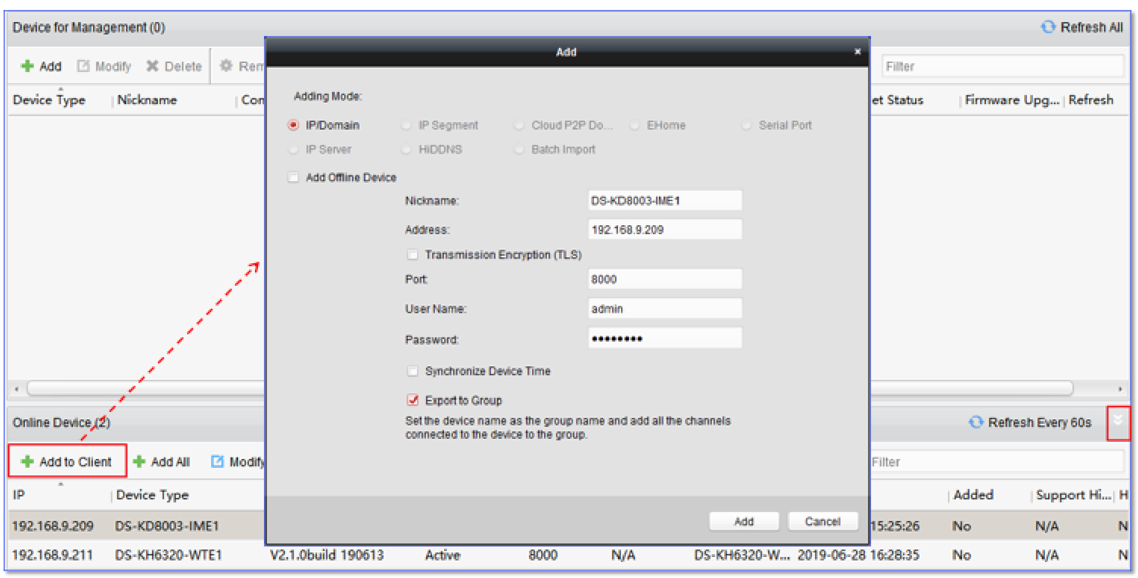
All active online devices in the same local subnet with the client software will be displayed on the **Online Device** area. Add video intercom devices to iVMS-4200 Client so that you can configure and manage them on the client.

Choose one of the following methods:

- **+ Add to Client**

  You can add devices manually and specify nickname for identification on iVMS-4200 client.

  1. Choose device and click **+ Add to Client**.
  2. Specify **Nickname** for the device and click **Add**.



  - **Nickname**: Enter a name for the device to identify. In this example, enter *DS-KD8003-IME1*.
  - **Address**: Enter IP address of the device. The IP address of the device is obtained automatically in this adding mode. In this example, IP address is *192.168.9.209*.
  - **Port**: Enter the device port number. The default value is **8000**.
  - **User Name**: Enter the user name to log in to the client. The default user name is **admin**.
  - **Password**: Enter the password to log in to the client. The default password is **12345**.

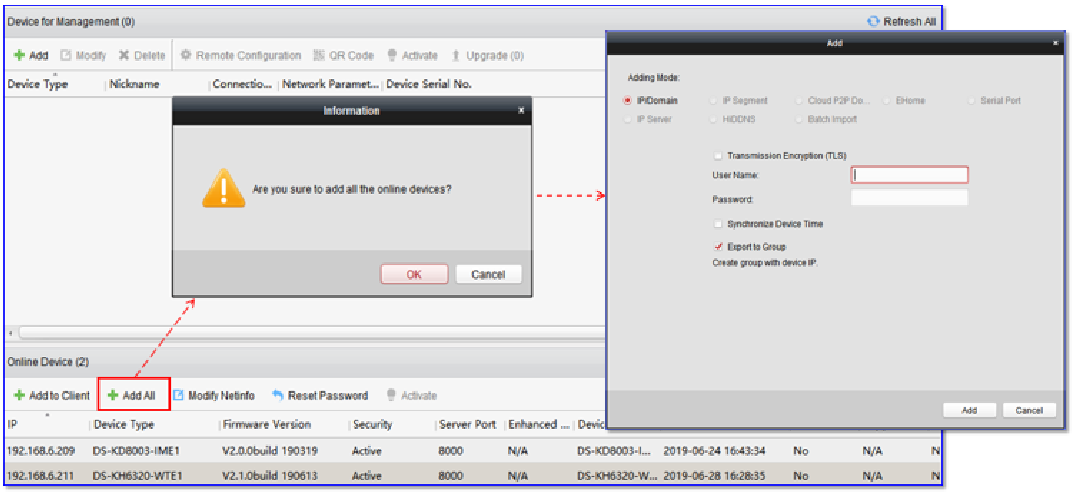
- **+ Add All**

  You can add all online devices to the client software with just one-click.

  1. Click **Add All**→ **OK**.
  2. Enter **User Name** and **Password**, and click **Add**. Then all available devices will be added to the iVMS-4200 Client with default setting.

     User Name: Enter the user name to log in to the client. The default user name is **admin**.

     Password: Enter the password to log in to the client. The default password is **12345**.



For more information about iVMS-4200 Client, refer to iVMS-4200 User Manual.

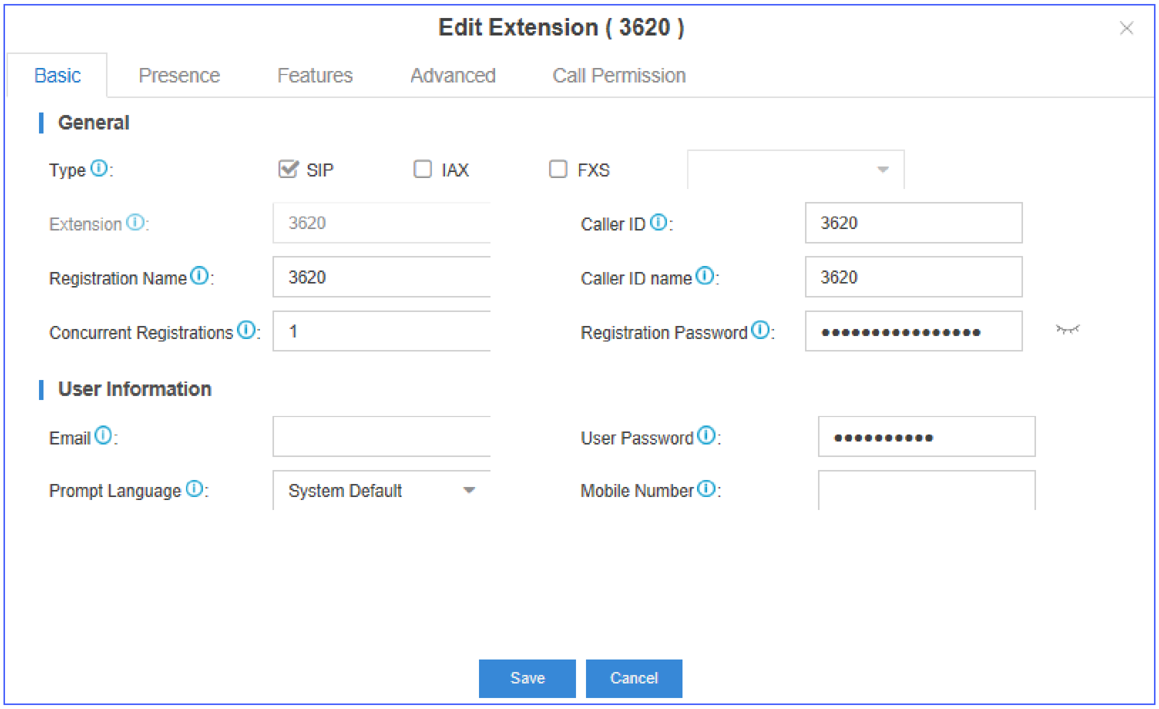
Step1. Configure Yeastar S-Series VoIP PBX

Before you start to configure the Hikvision devices, you need to add extensions for the Hikvision devices and configure the audio and video codecs on Yeastar S-Series VoIP PBX to ensure normal audio calls and video calls between the Hikvision devices.

Add two extensions for Hikvison devices

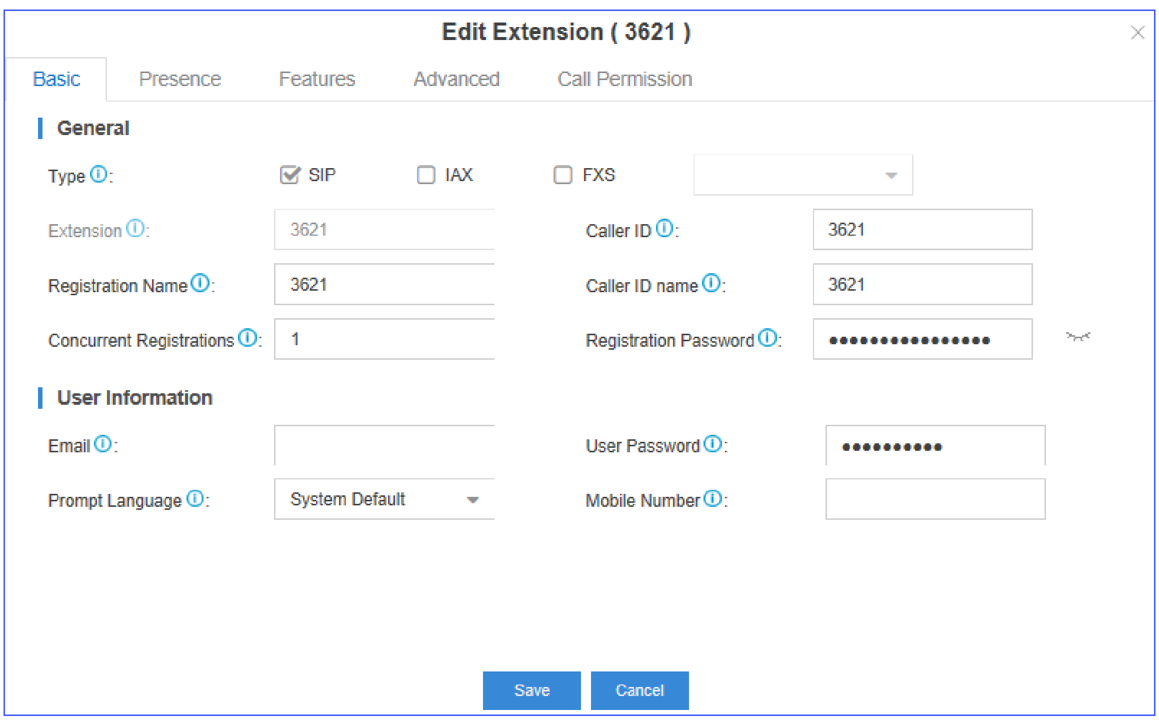
1. Log in the PBX web interface, go to **Settings**→ **PBX**→ **Extensions**, click **Add**.
2. Add an extension for Hikvision DS-KH6320 Indoor Station.

   In this example, add extension 3620 for Hikvision DS-KH6320 Indoor Station.



3. Add an extension for Hikvision DS-KD8003 Door Station.

   In this example, add extension 3621 for Hikvision DS-KD8003 Door Station.

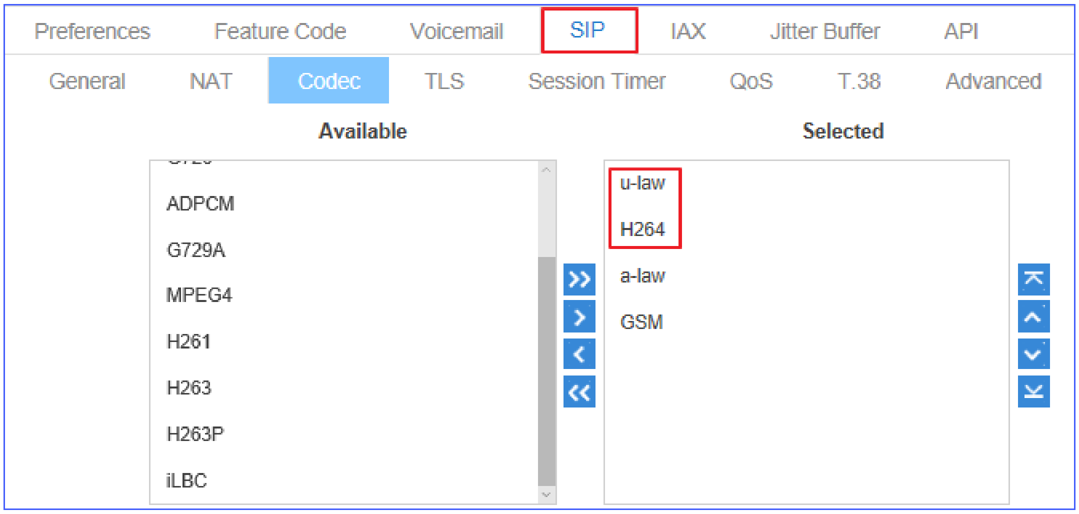


Configure audio and video codecs

The default audio and video codecs on Hikvision devices are **G711_U** and **STD_H264**.

To ensure the normal audio calls and video calls between Hikvision devices, the codecs **u-law** and **H264** should be selected on your PBX.

Go to **General**→ **SIP**→ **Codec**, check if the two codecs are selected.



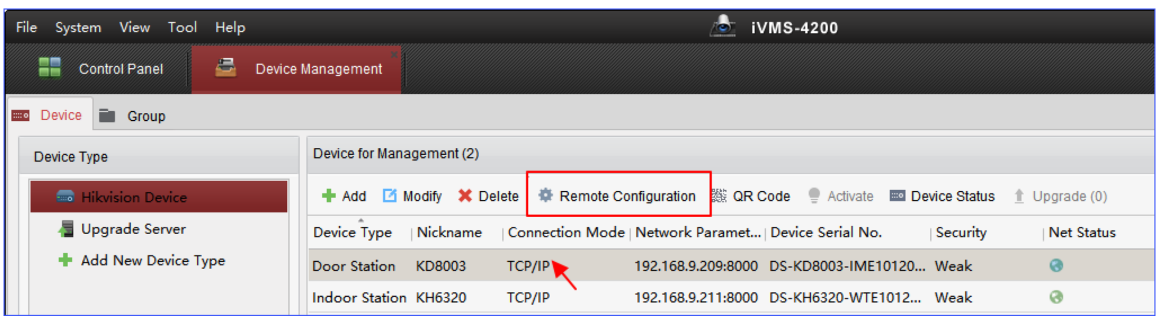
Step2. Configure Hikvision DS-KD8003 Door Station

Register an extension on Hikvision DS-KD8003 Door Station, configure the dial button, audio codecs, and video codecs.

Register an extension on DS-KD8003

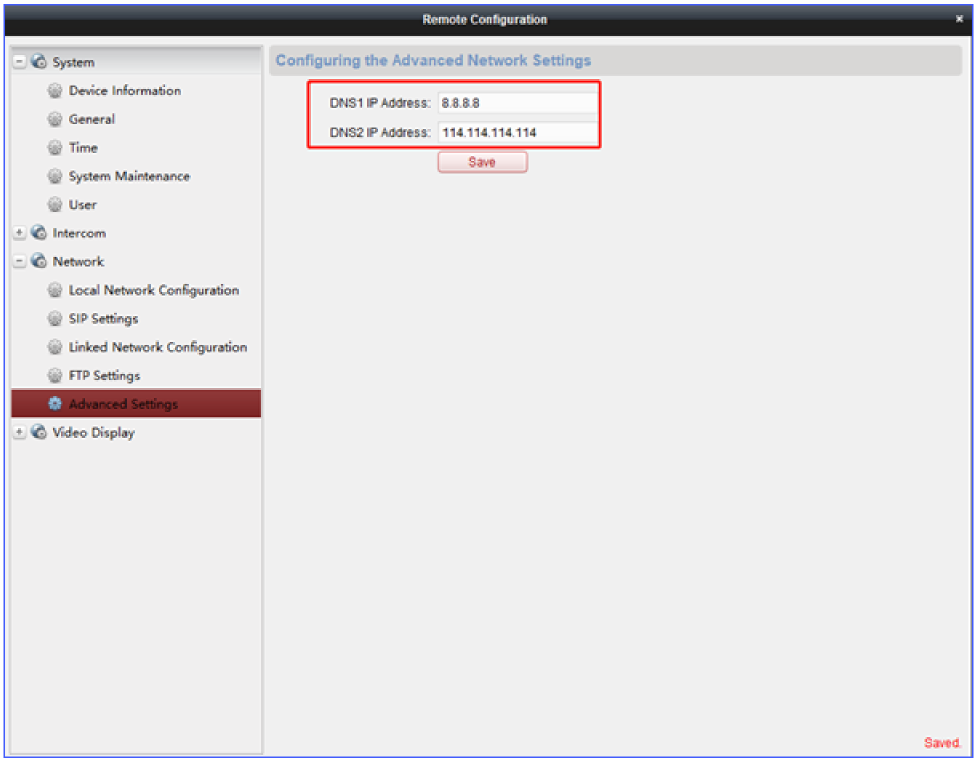
1. Log in iVMS-4200 client, go to **Device Management**→ **Device**→ **Hikvision Device**.

   Select DS-KD8003 Door Station and click **Remote Configuration**.

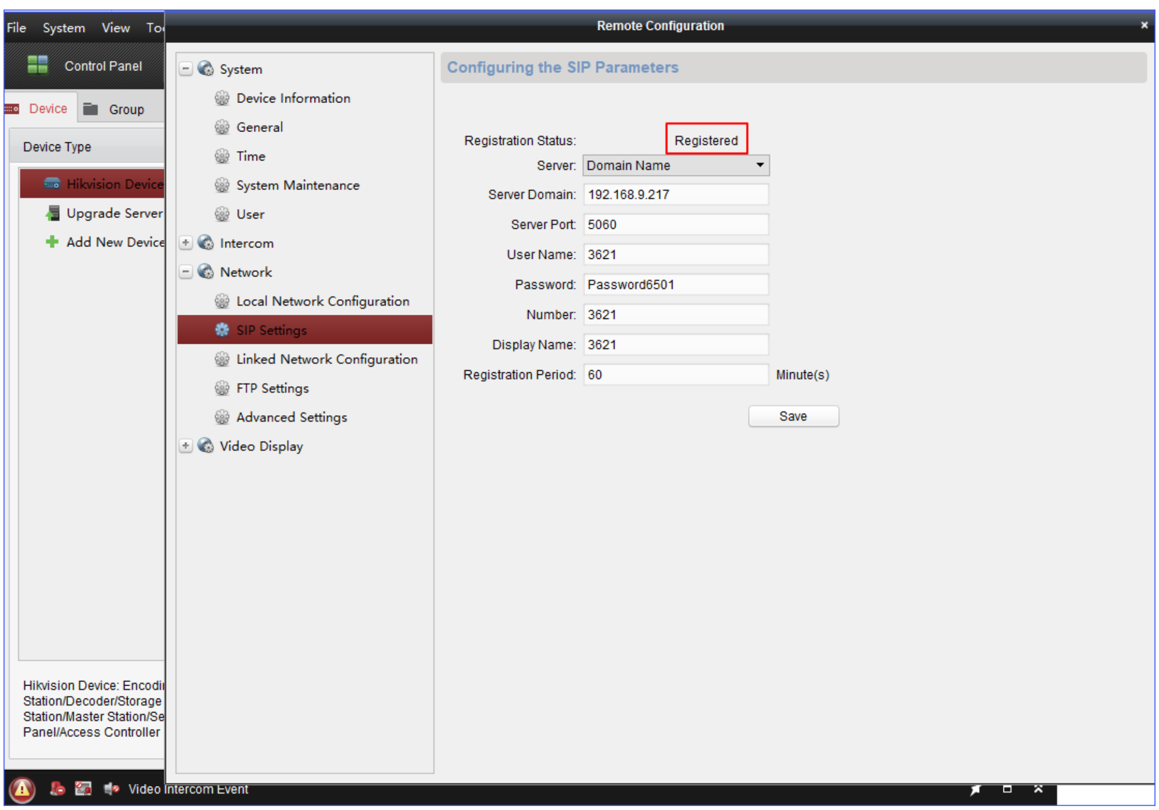


2. Go to **Intercom**→ **Intercom Protocol**, select **SIP Control** from the drop-down menu of **Protocol**, and click **Save**.
3. If your PBX is not in the same network with Hikvision DS-KD8003, you should configure DNS server.

   Go to **Network**→**Advanced Settings**, enter IP address of DNS server.



4. Go to **Network**→ **SIP Settings**, enter the credentials of SIP extension 3621.



- **Server Domain**: Enter IP address of Yeastar S100. In this example, enter *192.168.9.217*.

- **Server Port**: Enter SIP registration port of Yeastar S100. The default port is **5060**.
- **User Name**: Enter extension number. In this example, enter *3621*.
- **Password**: Enter registration password of the extension.
- **Number**: Enter extension number. In this example, enter *3621*.

5. Click **Save**.

   If the extension is registered, the **Registration Status** will display "Registered".

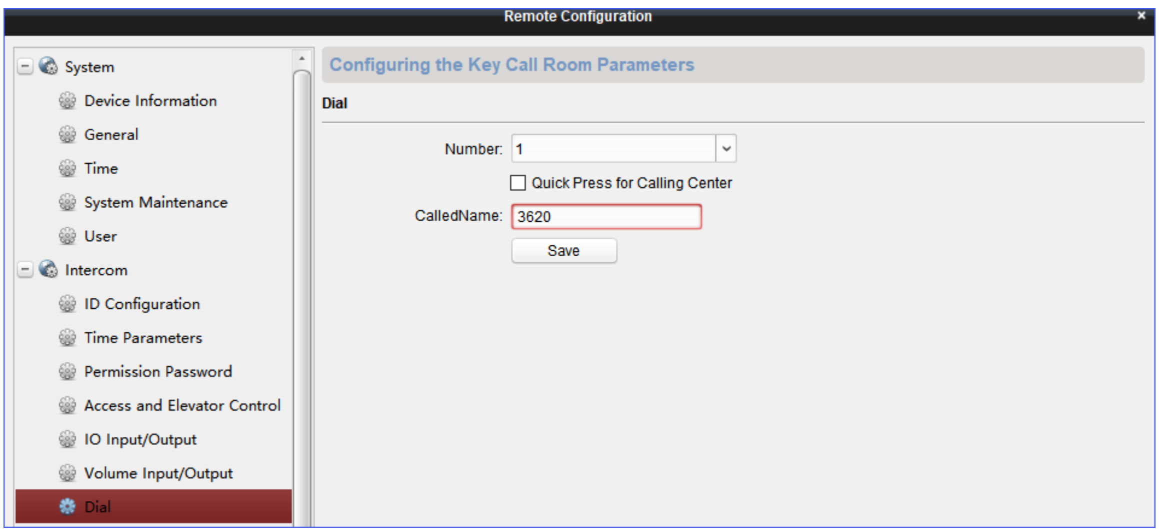
Dial Setting

1. On the **Remote Configuration** page of Hikvision DS-KD8003 Door Station, go to **Intercom→Dial**.
2. Configure the **Dial** settings:
   - **Quick Press for Calling Center**: Optional. This option is applied to a residential call center. When a guest presses the Dial button, the call will be received on iVMS-4200 client.

     Note: In our scenario, do NOT check this option, or the incoming calls cannot reach the DS-KH6320 Indoor Station.

   - **CalledName**: Enter the extension number of the DS-KH6320 Indoor Station. In this example, enter *3620*.

     When a guest presses the Dial button, the Indoor station will ring.

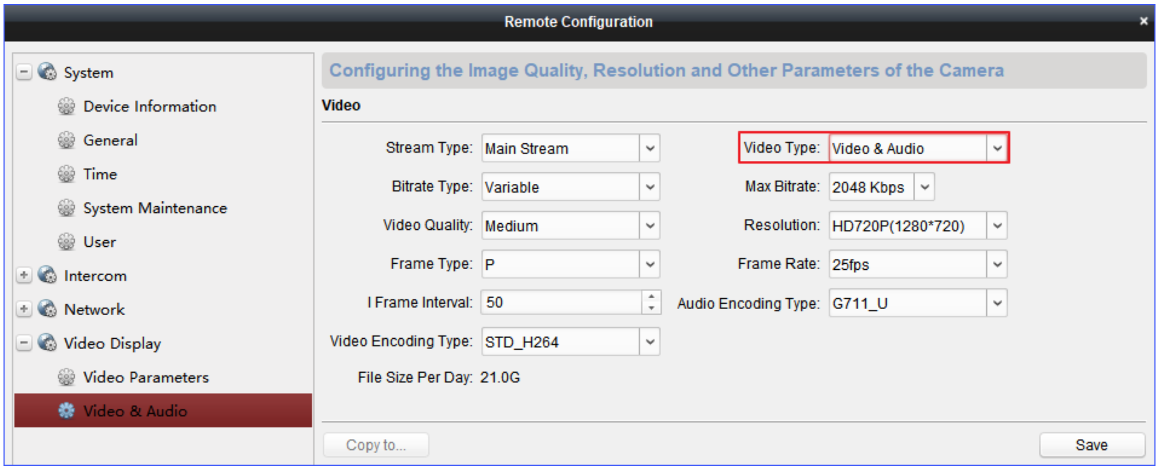


3. Click **Save**.

Audio & Video Setting

1. On the **Remote Configuration** page of Hikvision DS-KD8003 Door Station, go to **Video Display→ Video & Audio**, configure the following settings:
   - **Video Type**: Select **Video & Audio**, both video and two-way audio will be established when the call is answered.
   - **Audio Encoding Type**: G711_U
   - **Video Encoding Type**: STD_H264

   Note: Make sure the audio codec and video codec are selected on the PBX.



2. Click **Save**.

For more information about Hikvision DS-KD8003 Door Station, refer to DS-KD8003 Door Station User Guide.

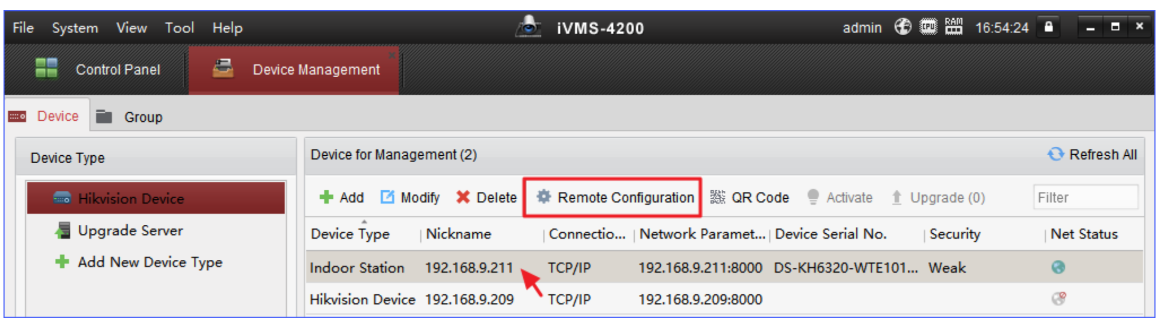
Step3. Configure Hikvision DS-KH6320 Indoor Station

Register an extension on Hikvision DS-KH6320 Indoor Station, and configure other settings of the Hikvision DS-KD8003 according to your usage scenarios.

Register an extension on DS-KH6320 Indoor Station

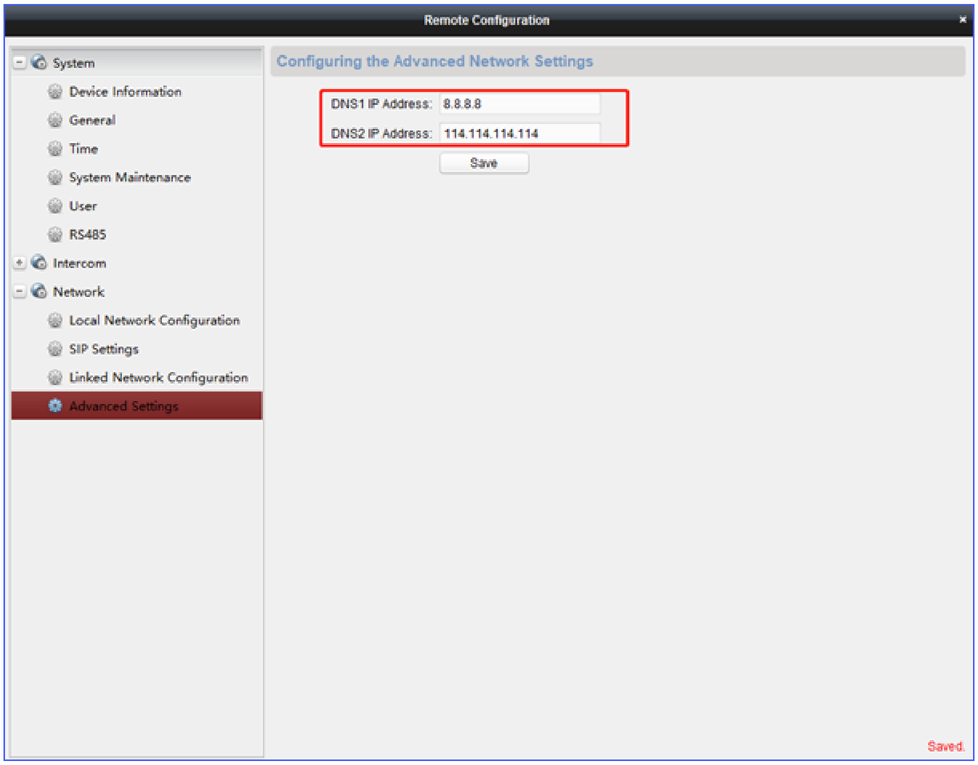
1. Log in to iVMS-4200 client, go to **Device Management**→ **Device**→ **Hikvision Device**.

   Select DS-KH6320 Indoor Station and click **Remote Configuration**.

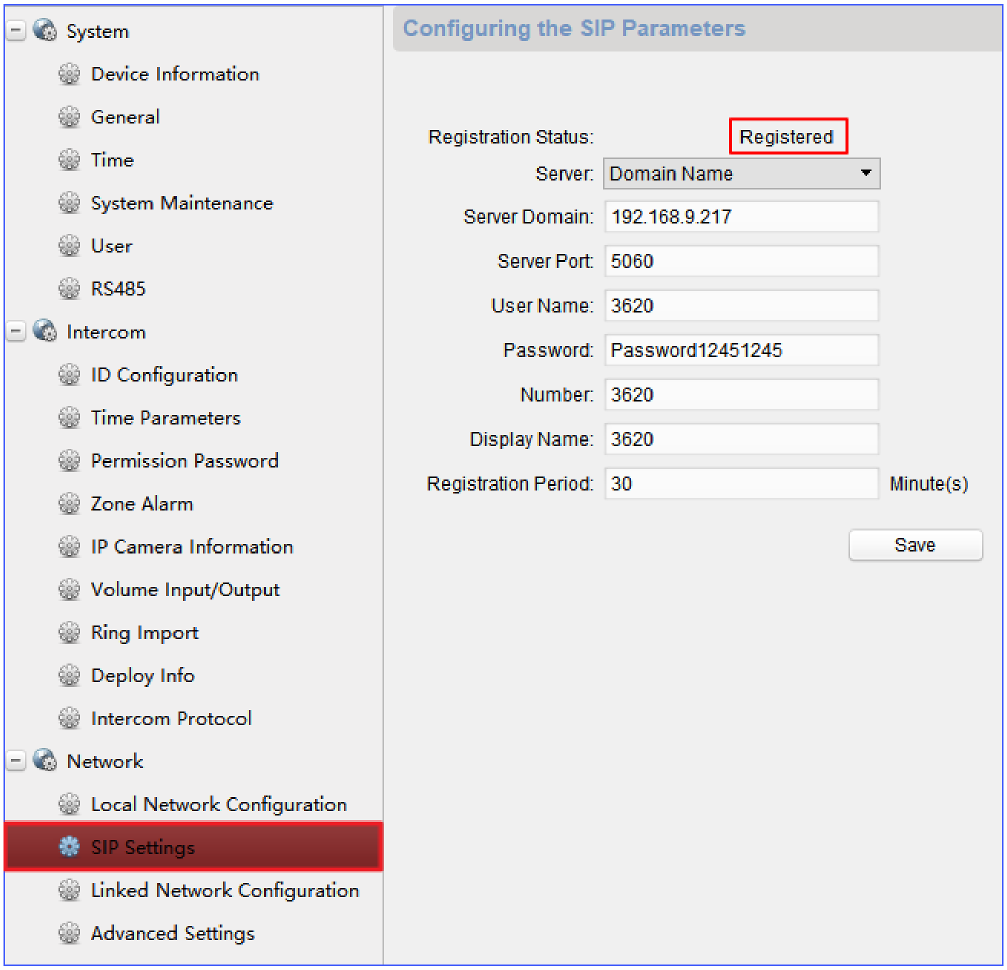


2. Go to **Intercom**→ **Intercom Protocol**, select **SIP Protocol** from the drop-down menu of **Protocol**, and click **Save**.
3. If your PBX is not in the same network with Hikvision DS-KH6320, you should configure DNS server.

   Go to **Network**→**Advanced Settings**, enter IP address of DNS server.



4. Go to **Network→ SIP Settings**, enter the credentials of SIP extension 3620, and click **Save**.



- **Server Domain**: Enter IP address of Yeastar S100. In this example, enter *192.168.9.217*.
- **Server Domain**: Enter IP address of Yeastar Cloud PBX. In this example, enter *yeastar.cloudpbx.com*.
- **Server Port**: Enter the SIP registration port of Yeastar S100. The default port is **5060**.
- **User Name**: Enter extension number. In this example, enter *3620*.
- **Password**: Enter registration password of the extension number.
- **Number**: Enter extension number. In this example, enter *3620*.

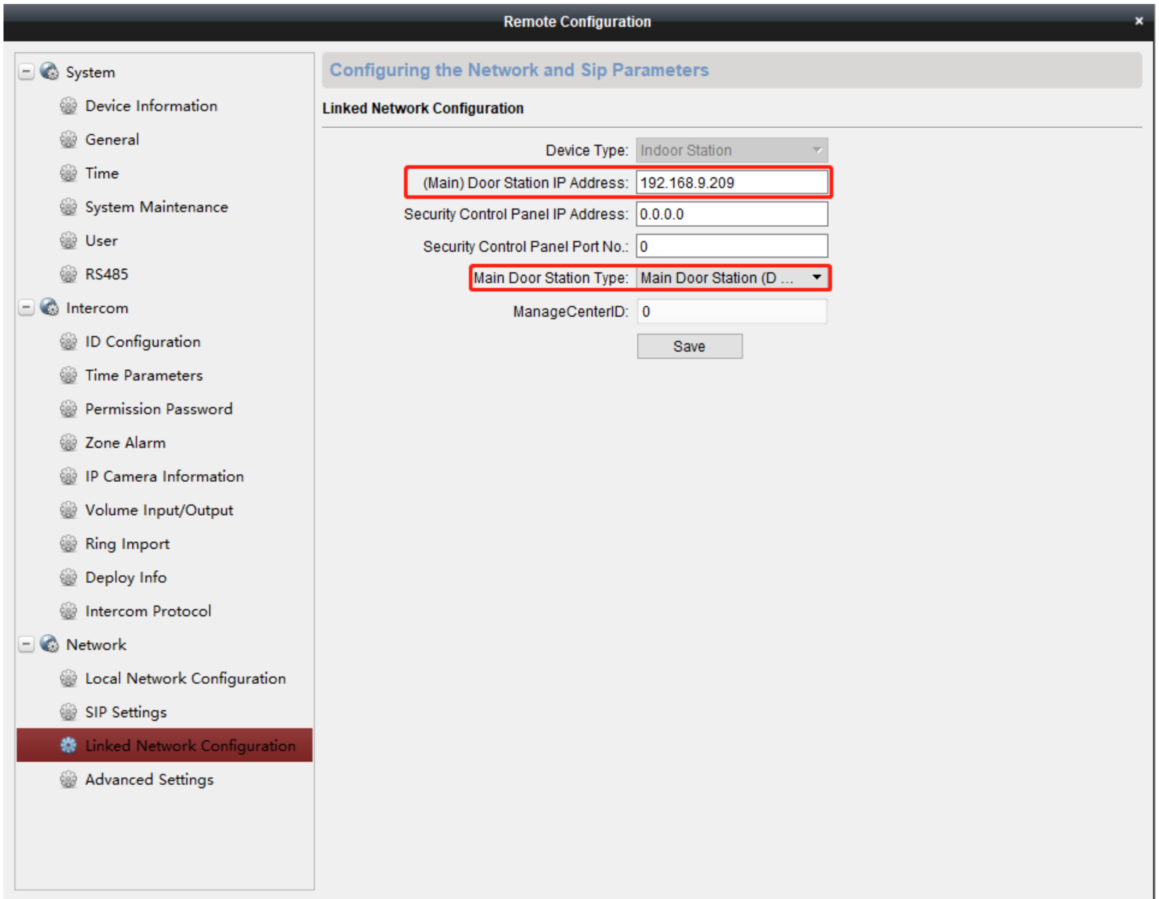
5. Click **Save**.

   If the extension is registered, the **Registration Status** will display "Registered".

6. Configure this Indoor Station to monitor the real-time status of the DS-KD8003 Door Station.

   Go to **Network→ Linked Network Configuration**, configure the following settings:

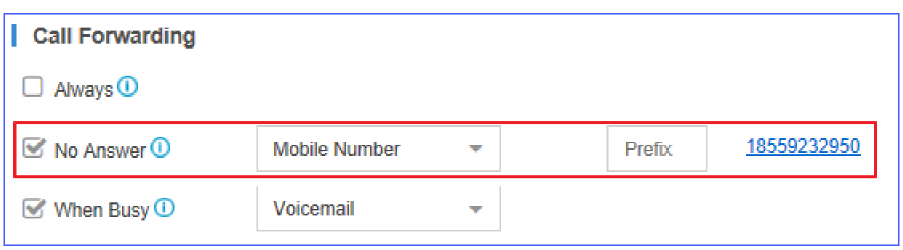
   - **(Main) Door Station IP Address**: Enter IP address of Hikvision DS-KD8003. In this example, enter *192.168.9.209*.
   - **Main Door Station Type**: Select **Main Door Station (D-series)**.



Call Forwarding settings

To prevent from missing any visits, you can set your mobile phone number as a destination of call forwarding. If no answer from the indoor station, your mobile phone will receive the call.

1. Configure Call Forwarding destination on Yeastar S100.
   a. Log in the PBX web interface, go to **Settings**→ **PBX**→ **Extensions**, edit the extension for Hikvision DS-KH6320 Indoor Station.
   b. Click **Presence** tab, configure **Call Forwarding** settings.

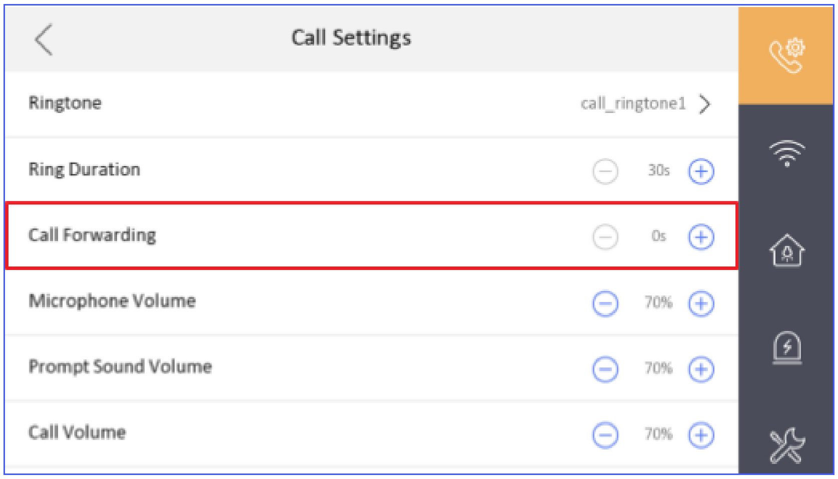


   - Select the checkbox **No Answer** and select the destination to **Mobile Number**.
   - Set Mobile Number.
   - Enter the **Prefix** according to the dial pattern settings of outbound route.

   c. Click **Save** and **Apply**.
2. Configure Call Forwarding time on Hikvision DS-KH6320 Indoor Station.
   a. On the Hikvision DS-KH6320, tap **Settings**→  to enter the **Call Settings** page.

b. Set the **Call Forwarding** time.

The ring duration limit beyond which the call is automatically forwarded to the mobile phone designated by the resident.



In this case, the call to 3620 (DS-KH6320 Indoor Station) will be forwarded to mobile phone after 30s. If no answer from mobile phone, the call will be dropped.

DND (Do Not Disturb) Settings

The DND (Do Not Disturb) feature allows you to set quiet hours for family time, movie time, or nap time. If DND is enabled on the Indoor Station, when a guest visits, the Indoor Station will not ring while the call log can be saved for your further query.

Note: DND settings on Hikvision DS-KH6320 Indoor Station has a higher priority over DND settings on the PBX.

Configure DND settings on Hikvision DS-KH6320:

1. On the Hikvision DS-KH6320, tap **Settings**→ to enter the **Call Settings** page.
2. Configure **Do Not Disturb** settings:
   - **Close**: The indoor station will ring every time it is called by door station or other indoor stations.
   - **All Day**: The indoor station will not ring all day when it is called by door station or other indoor stations, but the call logs will be saved.
   - **Schedule**: The indoor station will not ring between the start time and the end time when it is called by door station or other indoor stations, but the call logs will be saved.

For more information about Hikvision DS-KH6320 Indoor Station, refer to DS-KH6320 Indoor Station User Guide.

Htek

Register Htek Phone with Yeastar S-Series VoIP PBX

This article is based on Htek UC903 and Yeastar S-Series VoIP PBX v30.8.0.14.

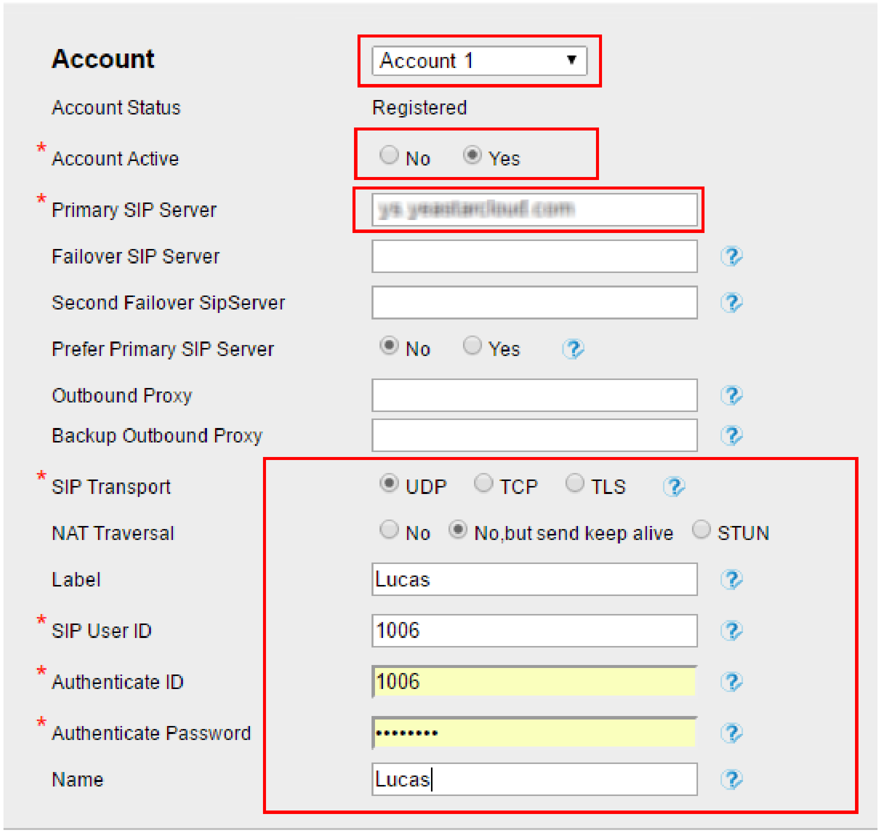
This article is applicable to the Htek UC series 802, 803, 804, 840, 842, 806, 862, 902, 903, 923, 924, 926.

Note: For the IP phone with different firmware version, the web GUI may be different.

1. Log in the web page of the phone.
   - **Username**: admin

- **Password**: admin

2. Click **Account** tab, choose one account to configure.



- **Account**: Select one account to configure.
- **Account Active**: Yes
- **Primary SIP Server**: Fill in the domain or IP address of your PBX.
- **SIP Transport**: Choose the same transport of the PBX. The default SIP transport on the PBX is UDP.
- **Label**: Set the name you want to appear on the phone screen.
- **SIP User ID**: Fill in the extension number.
- **Authentication ID**: Fill in the extension's **Registration Name**.
- **Authentication Password**: Fill in the extension's **Registration Password**.
- **Name**: The local phone name showing on the other phone when calling out.

3. Click **Save Set**.

   If the extension is registered, the page will show "Registered".

Panasonic

Register Panasonic Phone with Yeastar S-Series VoIP PBX

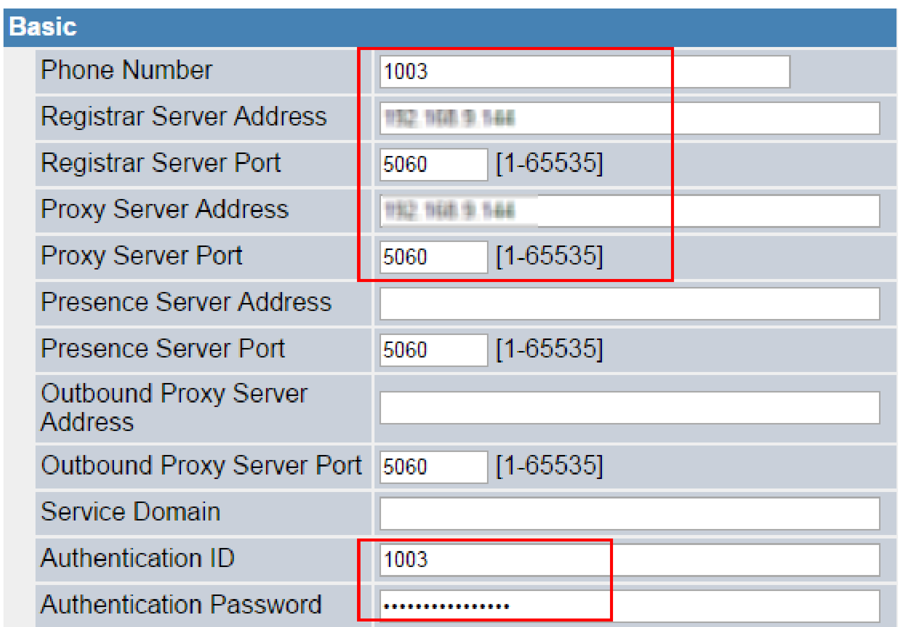
This article is based on Panasonic KX-HDV130 v01.008 and Yeastar S-Series VoIP PBX v30.8.0.14.

This article is applicable to the following Panasonic IP Phones.

- KX-HDV130
- KX-UT113

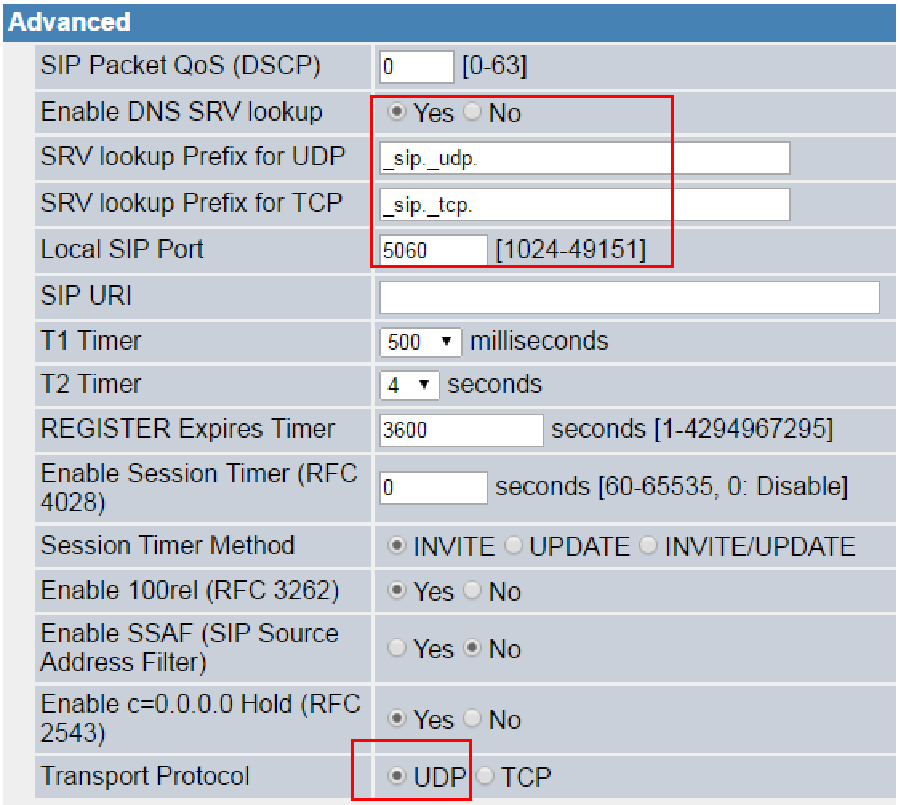
- KX-UT123
- KX-UT133
- KX-UT136
- KX-UT248
- KX-UT670
- TGP500
- TGP550

1. Start up the phone and check its IP address.
    a. Press **Menu**.
    b. Go to **System Settings**→**Network Settings**→**IPv4 Settings**→**Static**.
2. Open the web service for the Panasonic phone.
    a. Press **Menu**.
    b. Go to **Basic Settings**→**Other Option**→**Embedded Web**.
3. Log in the web page of the IP phone.
    - **Username**: admin
    - **Password**: adminpass
4. Click **VoIP**, choose a line to configure.
    a. In the **Basic** section:



    - **Phone Number**: Fill in the extension number.
    - **Registrar Server Address**: Fill the domain or IP address of your PBX.
    - **Registrar Server Port**: Fill in the SIP port of your PBX.
    - **Proxy Server Address**: Fill in the domain or IP address of your PBX.
    - **Proxy Server Port**: Fill in the SIP port of your PBX.
    - **Authentication ID**: Fill in the extension's **Registration Name**.
    - **Authentication Password**: Fill in the extension's **Registration Password**.

    b. In the **Advanced** section:



- **SRV lookup Prefix for UDP**: Enter `_sip_udp`.
- **SRV lookup Prefix for TCP**: Enter `_sip_tcp`.
- **Local SIP Port**: The SIP port number for each line must be unique, default value: 5060 (for Line 1) and 5070 (for Line 2).
- **Transport Protocol**: Choose the same transport protocol as the PBX.

5. Click **Save**.

   If the extension is registered, you can see the VoIP status shows "Registered".

Polycom

Register Polycom Phone with Yeastar S-Series VoIP PBX

This guide is based on Polycom VVX 201 and Yeastar S-Series VoIP PBX v30.8.0.14.

This guide is applicable to the following phones:

- Polycom VVX Series: 101, 201, 300, 310, 400, 500, 600, 601, 1500
- Polycom SoundPoint Series: IP321, IP331, IP335, IP450, IP550, IP560, IP670

Note: For the IP phone with different firmware version, the web GUI may be different.

1. To check the IP address of the phone, press **Menu** on the phone, go to **Settings→Status→Network→TCP/IP Parameter**.
2. Enable Web service for the phone.
   a. Press **Menu** on the phone, go to **Settings→Advanced**, enter the password 456.
   b. Go to **Administration Settings→Web Server Configuration**, configure the following:

- **Web Server**: Enabled
- **Web Config Mode**: Choose HTTP or HTTPS

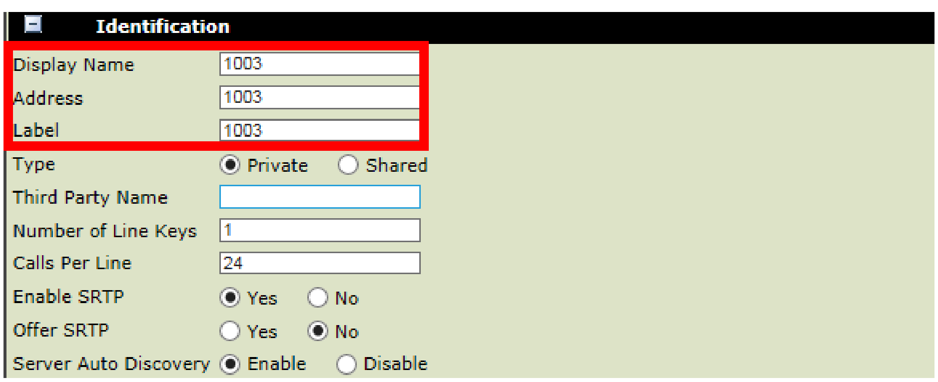
3. Log in the web page of the phone.

Note: For the firmware version 5.5.0 or later, the phone only supports HTTPS web login. You need use HTTPS to log in the web page. For example, type `https://192.168.6.160` in your web browser to access the phone web page.

- **Login as**: Admin
- **Password**: 456

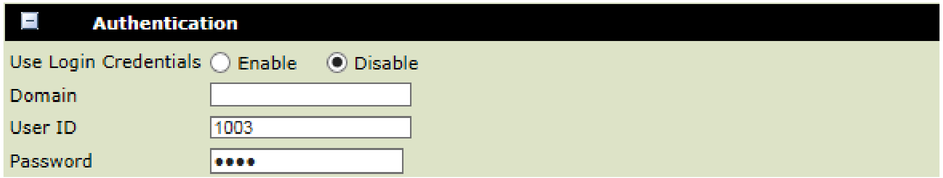
4. Go to **Settings→Lines**, choose a line to configure.
   a. Enable **SIP Protocol**.

   b. Expand **Identification** option, and set as the following:



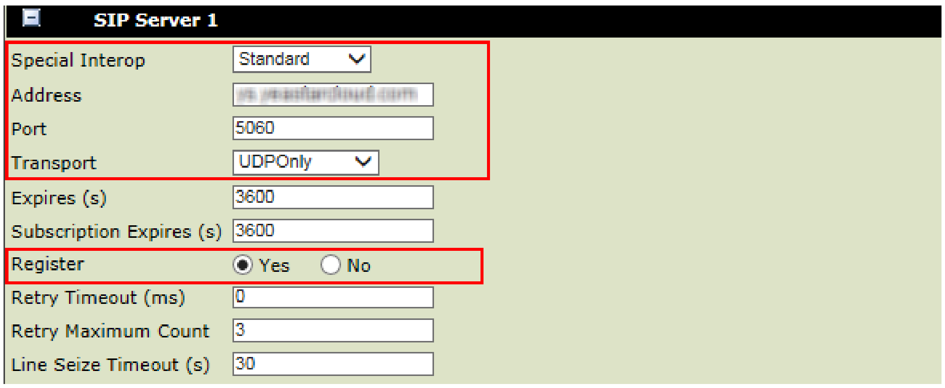
- **Display Name**: Set the name you want to appear on other phone's screen when calling other phones.
- **Address**: Fill in the extension number.
- **Label**: Set the name you want to appear on the phone screen.

   c. Expand **Authentication** option, and set as the following:



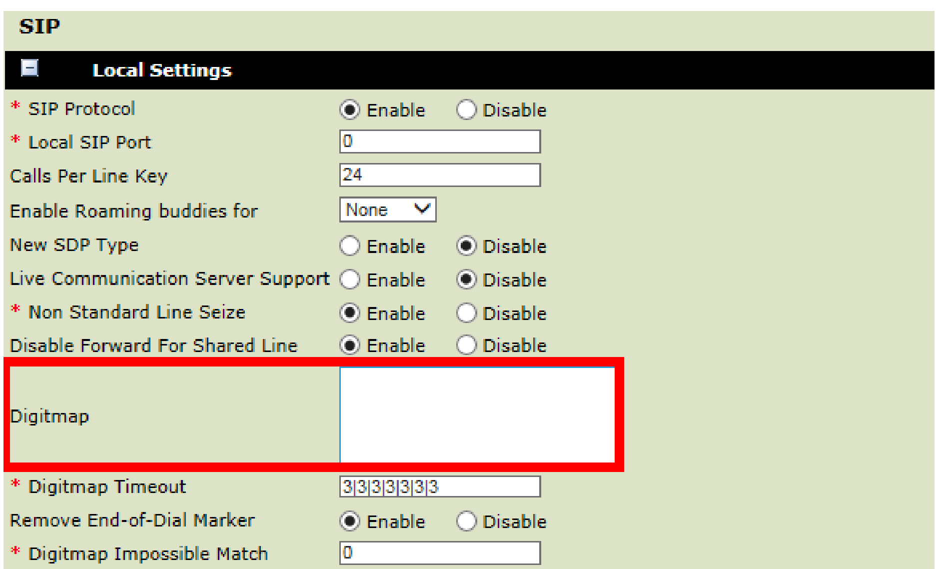
- **User Login Credentials**: Disable
- **User ID**: Fill in the extension's **Registration Name**.
- **Password**: Fill in the extension's **Registration Password**.

d. Expand **SIP Server 1** option, and set as the followings:



- **Special Interop**: Standard
- **Address**: Fill in the domain or IP address of your PBX.
- **Port**: Fill in the same SIP port as the PBX.
- **Transport**: Choose the same transport protocol as the PBX.
- **Register**: Yes

5. Go to **Settings→SIP**, set the **Digitmap** to blank. In this way, you can dial any number out.



6. Click **Save**.

If the extension is registered, the **Register Status** will show "Registered".

Snom

Register Snom Phone with Yeastar S-Series VoIP PBX

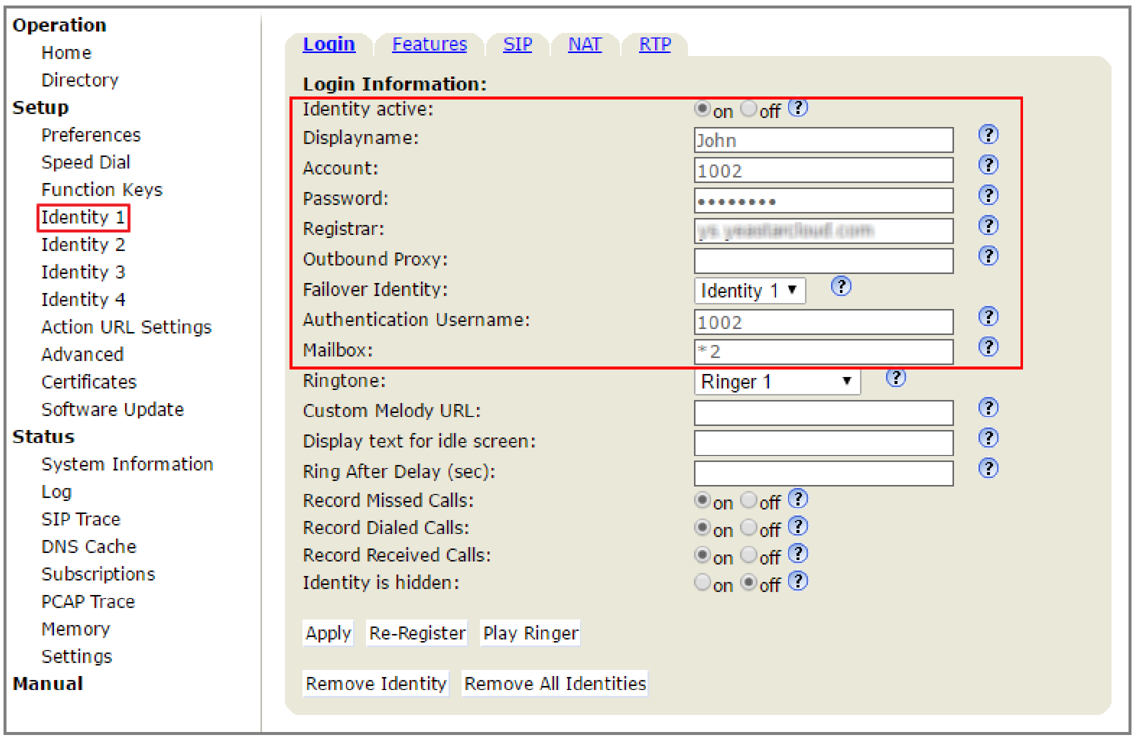
This article is based on Snom D305 and Yeastar S-Series VoIP PBX v30.8.0.14.

This article is applicable to the following phones:

- Snom 320, 710, 715, 720, 725, 760, 765
- Snom D Series: 305, 315, 345, 375

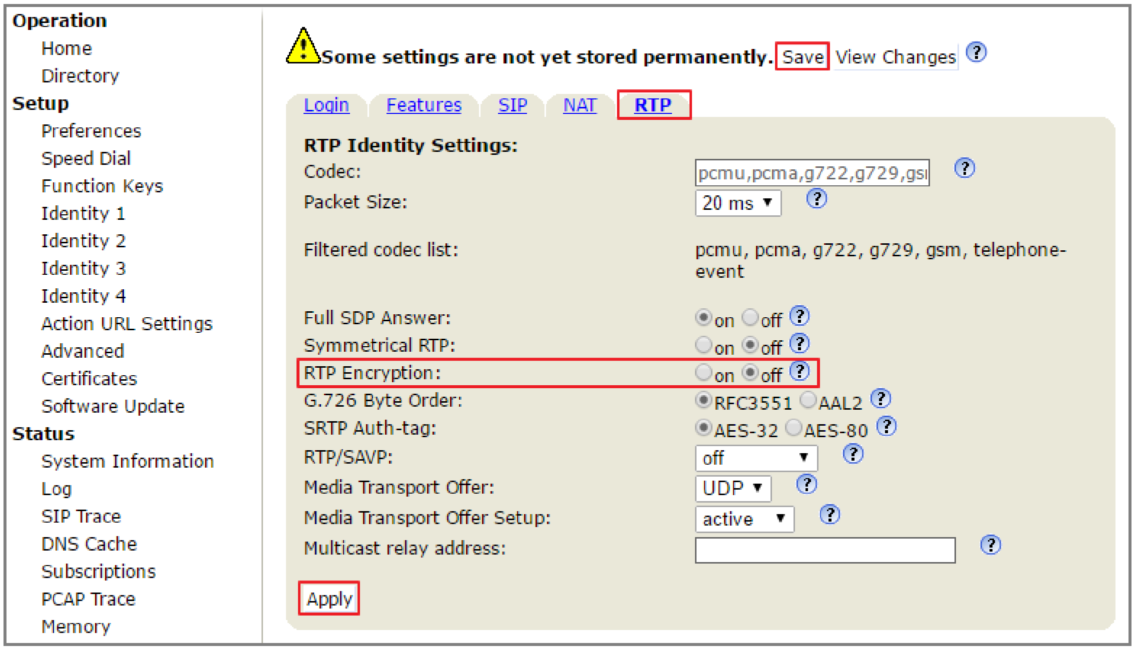
Note: For the IP phone with different firmware version, the web GUI may be different.

1. To check the IP address of the phone, press **Settings→Information→System Info** or press **Menu→Information→System Info**.
2. Log in the web page of the phone, go to **Setup→Identify 1** to configure the account 1.



- **Identify active**: On
- **Displayname**: Fill in the name you wish to appear on the phone screen.
- **Account**: Fill in the extension number.
- **Password**: Fill in the extension's **Registration Password**.
- **Registrar**: Fill in the domain or IP address of your PBX.
- **Authentication Username**: Fill in the extension's **Registration Name**.
- **Mailbox**: Fill in the feature code of **Check Voicemail** on the PBX. The default code is *2.

3. Click **RTP** tab, set **RTP Encryption** to **Off** if you don't enable SRTP feature for the extension.



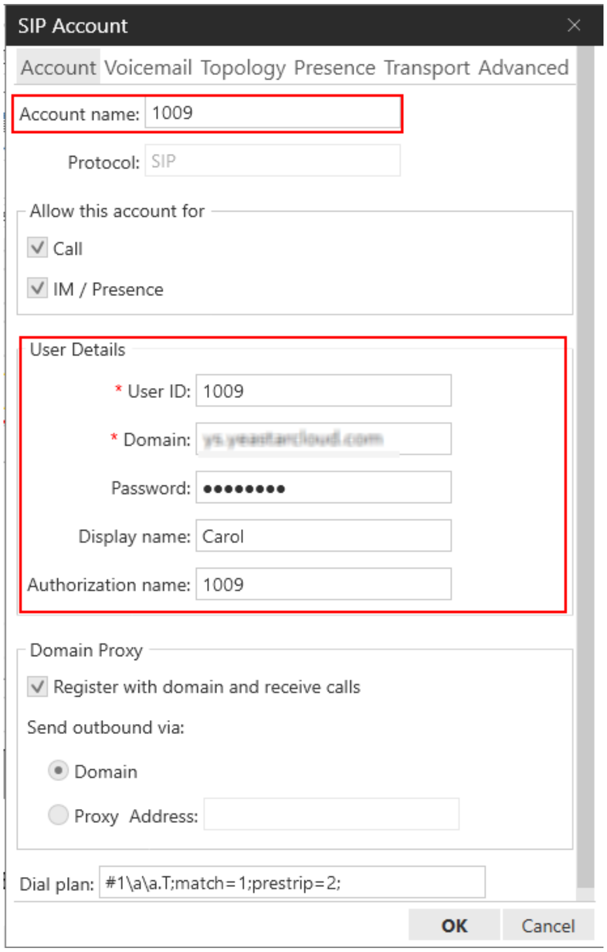
4. Click **Apply**, then click **Save** in the top-right corner.

X-Lite

Register X-Lite Soft Phone with Yeastar S-Series VoIP PBX

This guide is based on X-Lite PC client v5.2.0 and Yeastar S-Series VoIP PBX v30.8.0.14.

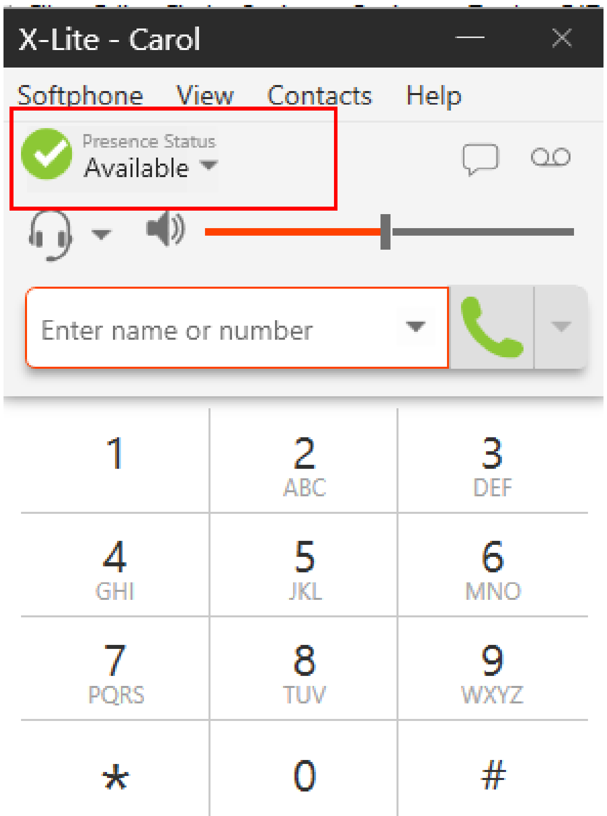
1. Launch X-Lite, go to **Softphone→Account Settings**, configure the SIP account.



- **Account name**: Set a name for the account.
- **User ID**: Enter the extension number.
- **Domain**: Enter the domain or IP address of your PBX.
- **Password**: Enter the extension's **Registration Password**.
- **Display name**: Set the name that you want to appear on the soft phone screen.
- **Authorization name**: Enter the extension's **Registration Name**.

2. Click **OK**.

If the extension is registered, you can see the status shows as below.

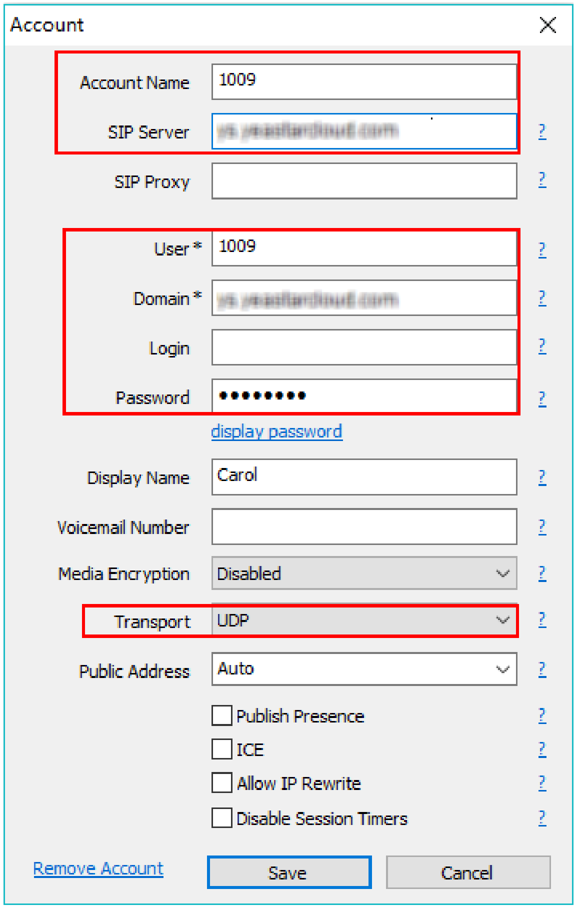


MicroSIP

Register MicroSIP Soft Phone with Yeastar S-Series VoIP PBX

This guide is based on the MicroSIP v3.17.3 and Yeastar S-Series VoIP PBX v30.8.0.14.

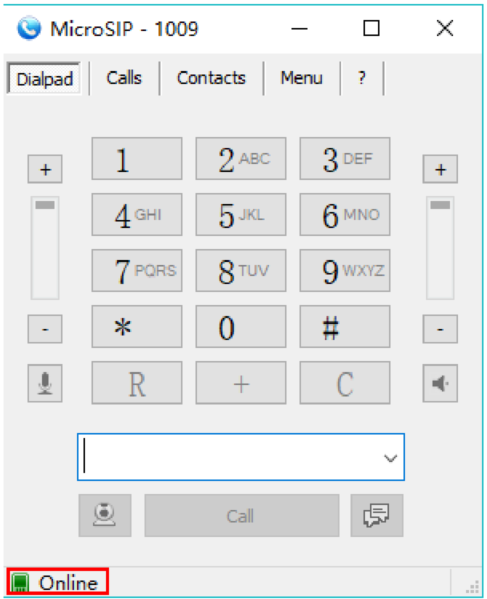
1. Launch MicroSIP, go to **Menu**→**Add Account**, configure the account settings.



- **Account Name**: Set the name that you want to appear on the soft phone screen.
- **SIP Server**: Enter the domain or IP address of your PBX.
- **User**: Enter the extension number.
- **Domain**: Enter the domain or IP address of your PBX.
- **Password**: Enter the extension's **Registration Password**.
- **Display Name**: Set the name you want to appear on the other phone's screen when calling out.
- **Transport**: Choose the same protocol of the PBX. The default protocol on PBX is UDP.

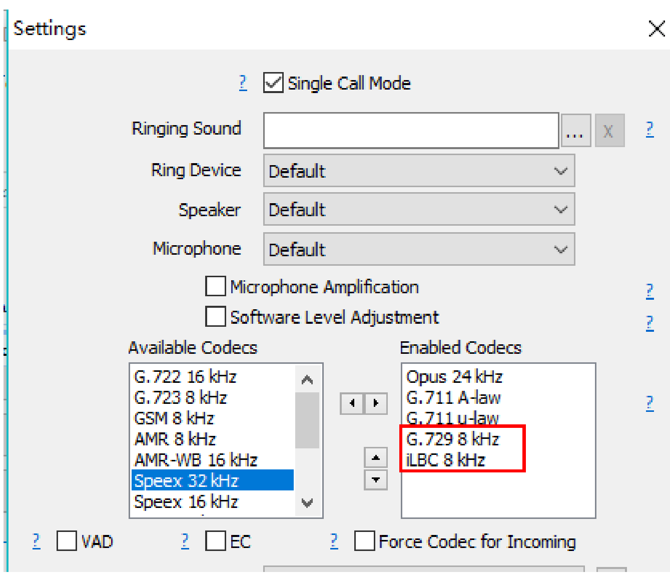
2. Click **Save**.

   If the extension is registered, you can see the status shows as below.



3. Go to **Menu**→**Settings**, enable G729 and iLBC codecs.

   Note: G729 and iLBC are the default enabled codecs on the PBX. To ensure the call is normal, you need to enable the G729 or iLBC codec on the soft phone.



4. Click **Save**.

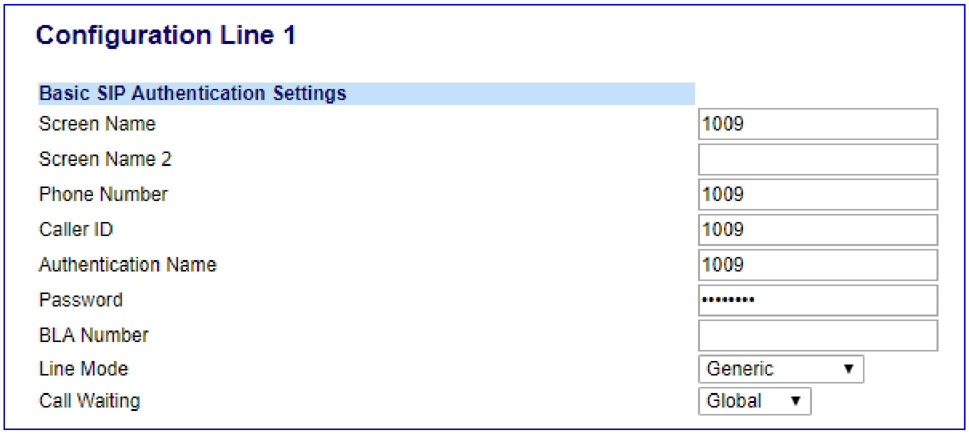
Mitel

Register Mitel Phone with Yeastar S-Series VoIP PBX

This guide is based on Mitel 6867i v4.1.0.148 and Yeastar S-Series VoIP PBX v30.8.0.14.

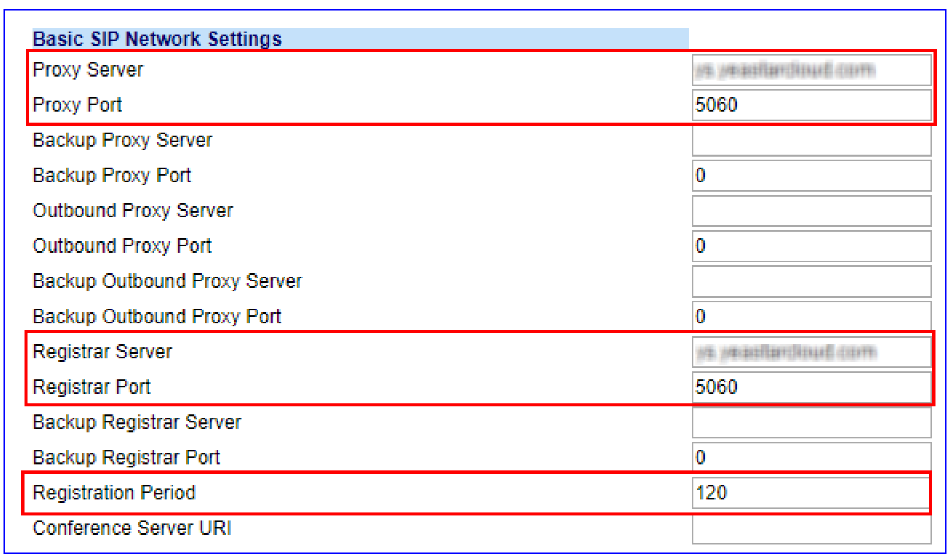
Note: For the IP phone with different firmware version, the web GUI may be different.

1. Log in the web page of the phone.
    - **Username**: admin
    - **Password**: 22222
2. Go to **Advanced** section, choose a line to configure.
    a. In the **Basic SIP Authentication Settings**, enter the extension information.



- **Screen Name**: Set the name that you want to display on the phone screen.
- **Phone Number**: Fill in the extension number.
- **Caller ID**: Fill in the extension's **Caller ID**.
- **Authentication Name**: Fill in the extension's **Registration Name**.
- **Password**: Fill in the extension's **Registration Password**.

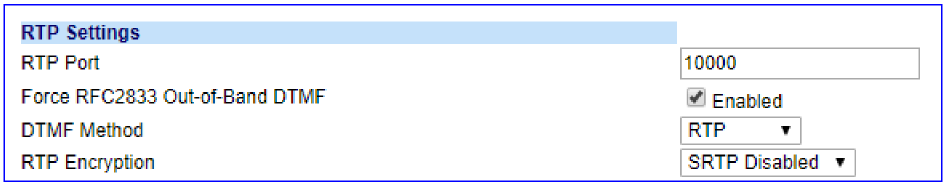
b. In the **Basic SIP Network Settings**, enter the PBX information.



- **Proxy Server**: Fill in the domain or IP address of your PBX.
- **Proxy Port**: Fill in the same SIP port of the PBX. The default SIP port on the PBX is 5060.
- **Registrar Server**: Fill in the domain or IP address of your PBX.
- **Registrar Port**: Fill in the same SIP port of the PBX. The default SIP port on the PBX is 5060.
- **Registration Period**: Set the registration period according to the settings on your PBX. The default range of SIP registration time on the PBX is 60-3600 seconds.

3. Click **Save Settings**.

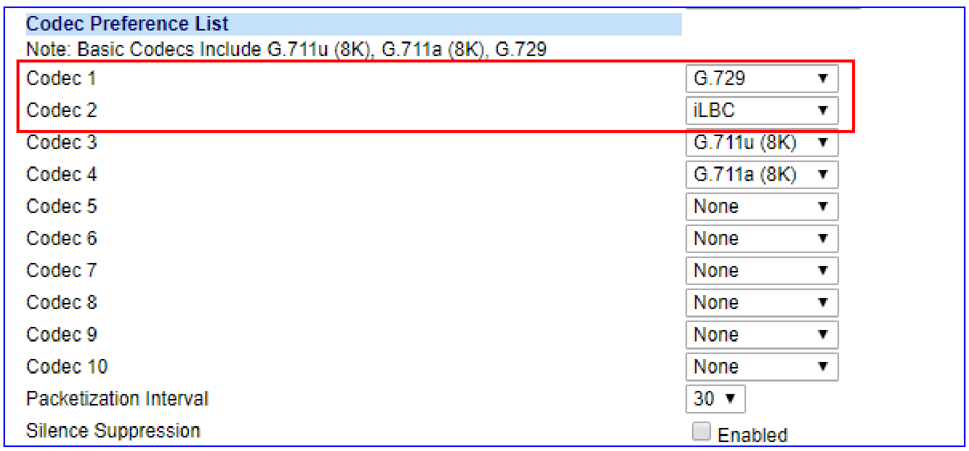
4. Go to **Advanced→Global SIP**, set the RTP settings and codec preferences.
   a. In the **RTP Settings** section, configure the RTP according to the settings on your PBX.



      - **Force RFC2833 Out-of-Band DTMF**: Enabled
      - **DTMF Method**: RTP
      - **RTP Encryption**: If you don't enable SRTP for the extension, choose **SRTP Disabled**.

   b. In the **Codec Preference List** section, set the codec preferences according your PBX settings.

Note: G729 and iLBC are the default enabled codecs on the PBX, you should enable the G729 codec or the iLBC codec on your phone.



5. Click **Save Settings**.
6. Reboot the phone to make the configuration take effect.

   You can check the extension status via **Status→System Information**. If the extension is registered, the status shows "Registered".

Vtech

Register Vtech Phone with Yeastar S-Series VoIP PBX

This guide is based on the Vtech VSP610A v2. 0. 3. 0 and Yeastar S-Series VoIP PBX v30.8.0.14.

Note: For the IP phone with different firmware version, the web GUI may be different.

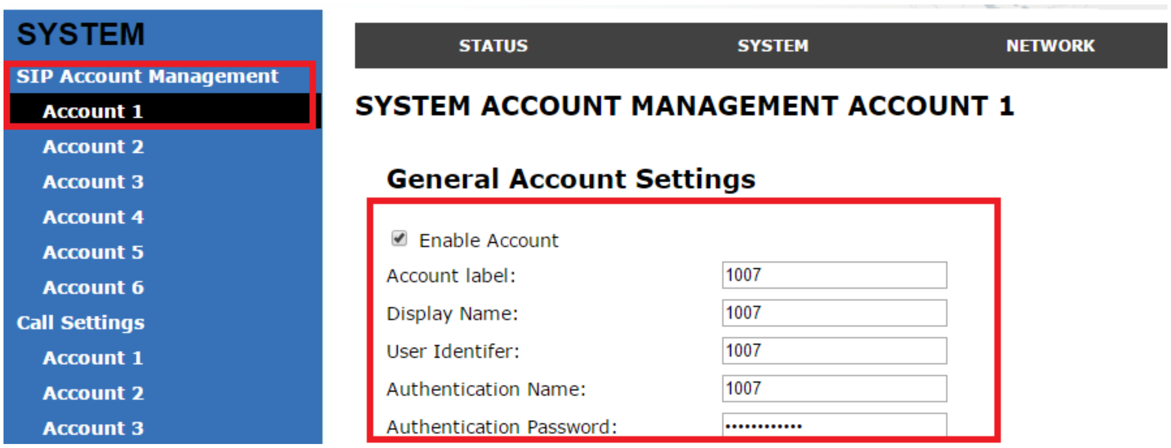
Configure the IP address via phone user interface

1. Press **System→Network→Basic Settings→Dual Mode→WAN Setting**.
2. Choose **Static IP** and alter the **IP Address, Subnet Mask**, **Preferred DNS Server**, **Alternate DNS Server**.
3. **Apply** it after input the correct information.
4. **Reboot** the phone and log in the phone web user interface using the new IP address.
5. Enter the user name and password, click **Log In** to enter the web user interface.

- **User Name**: admin
- **Default Password**: admin

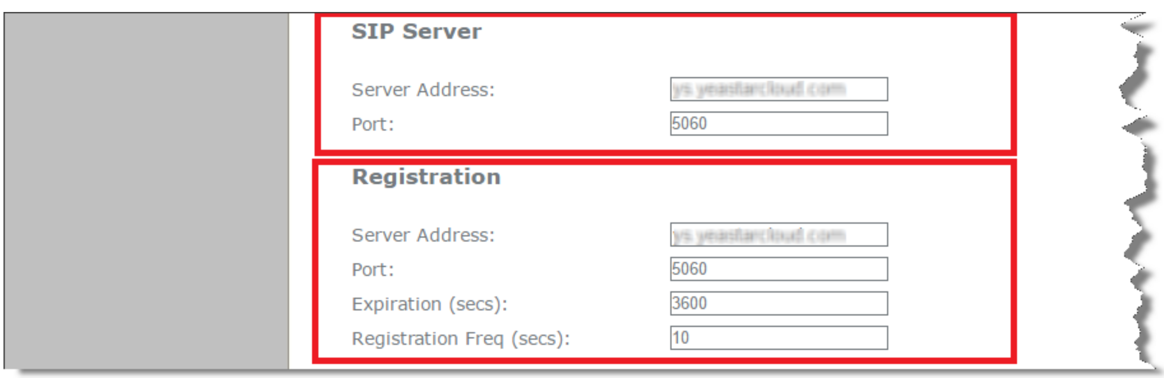
Account Registration

1. Log in the IP phone, go to **System→SIP Account Management**, select one account to configure.
2. Enable the account and fill in the extension information.



- **Enable Register**: check
- **Account Label**: The name you want to display on the phone screen.
- **Display Name**: The name you want to display on another person's phone screen when you are calling the phone.
- **User Identifier**: Enter the extension's **Caller ID**.
- **Authentication Name**: Enter the extension's **Registration Name**.
- **Authentication Password**: Enter the extension's **Registration Password**.

3. In the **SIP Server** section and **Registration** section, fill in your PBX information.



- **SIP Server**
  - **Server Address:** Enter the domain or IP address of your PBX.
  - **Server Port:** Enter the SIP port of your PBX.
- **Registration**
  - **Server Address:** Enter the domain or IP address of your PBX.

- **Port:** Enter the SIP port of your PBX.

4. Click **Apply**.

   If the registration is successfully, the register status would show "Registered".

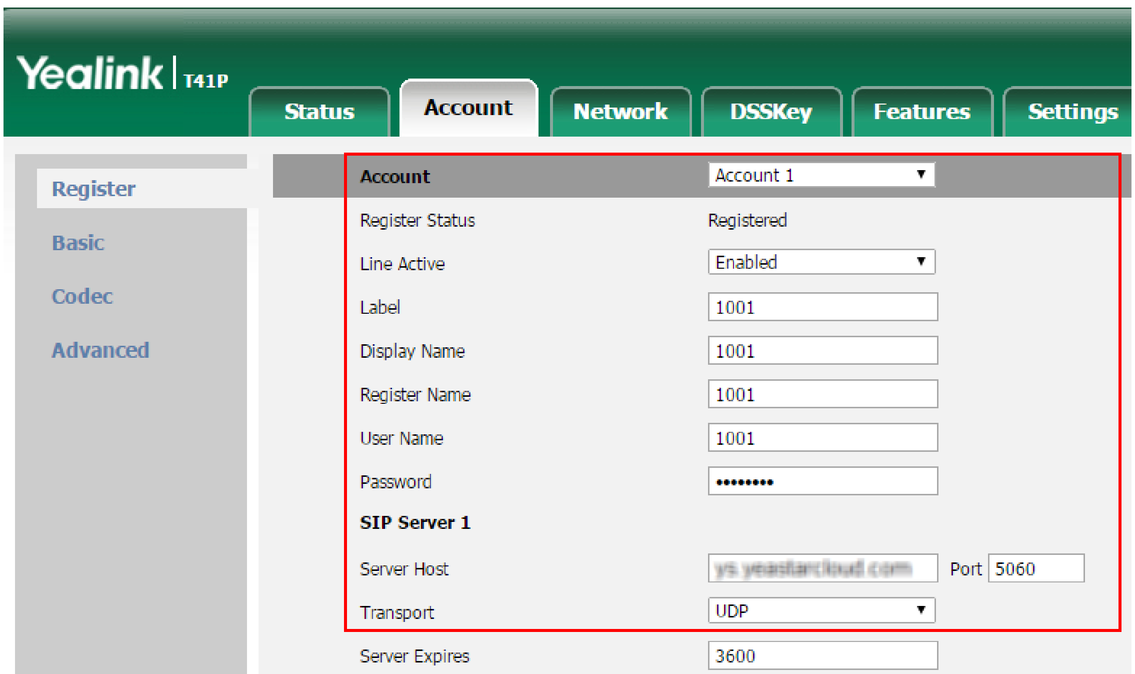
Yealink

Register Yealink Phone with Yeastar S-Series VoIP PBX

This guide is applicable to all the Yealink phones and Yeastar S-Series VoIP PBX v30.8.0.14.

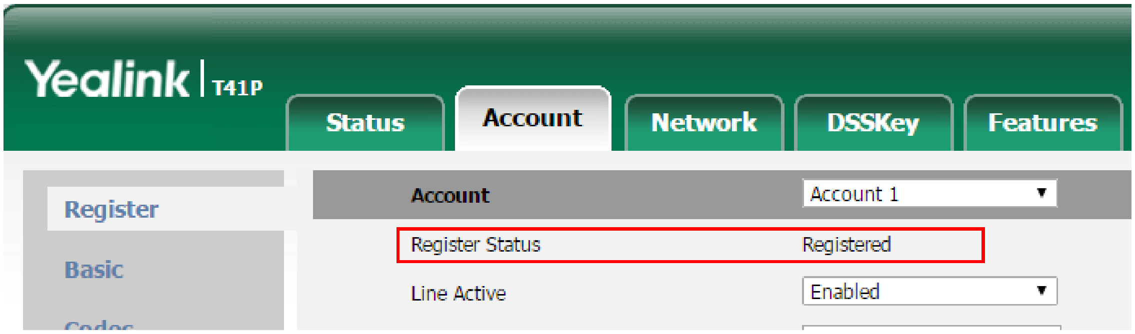
Note: For the IP phone with different firmware version, the web GUI may be different.

1. Log in the web page of the phone.
   - **Username**: admin
   - **Password**: admin
2. Click **Account** tab, and choose one account to configure.



   - **Account**: Choose one account.
   - **Line Active**: Enabled
   - **Label**: Set the name you want to appear on the phone screen.
   - **Display Name**: Set the name you want to appear on the other phone's screen when calling out.
   - **Register Name**: Fill in the extension's **Register Name**.
   - **User Name**: Fill in the extension number.
   - **Password**: Fill in the extension's **Registration Password**.
   - **Server Host**: Fill in the domain or IP address of your PBX.
   - **Port**: Fill in the same SIP port of the PBX.
   - **Transport**: Choose the same transport protocol of your PBX.
3. Click **Confirm**.

If the extension is registered, you can see the **Register Status** shows "Registered".

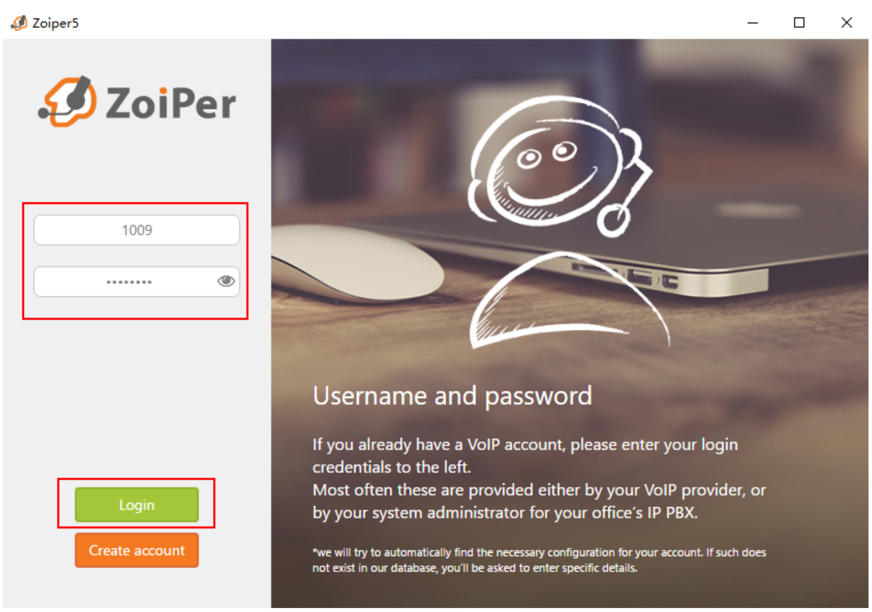


Zoiper

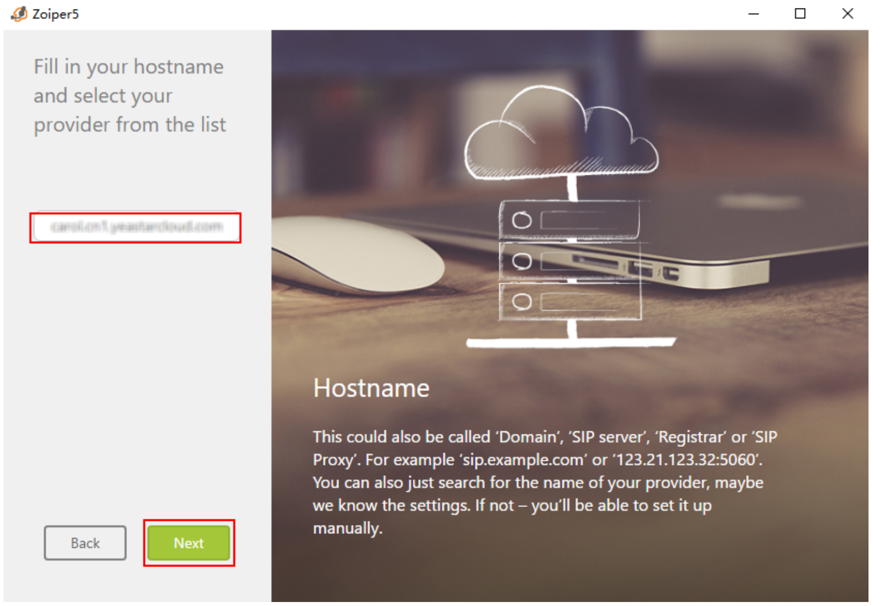
Register Zoiper Soft Phone with Yeastar S-Series VoIP PBX

This guide is based on the Zoiper PC client v5.2.12 and Yeastar S-Series VoIP PBX v30.8.0.14.

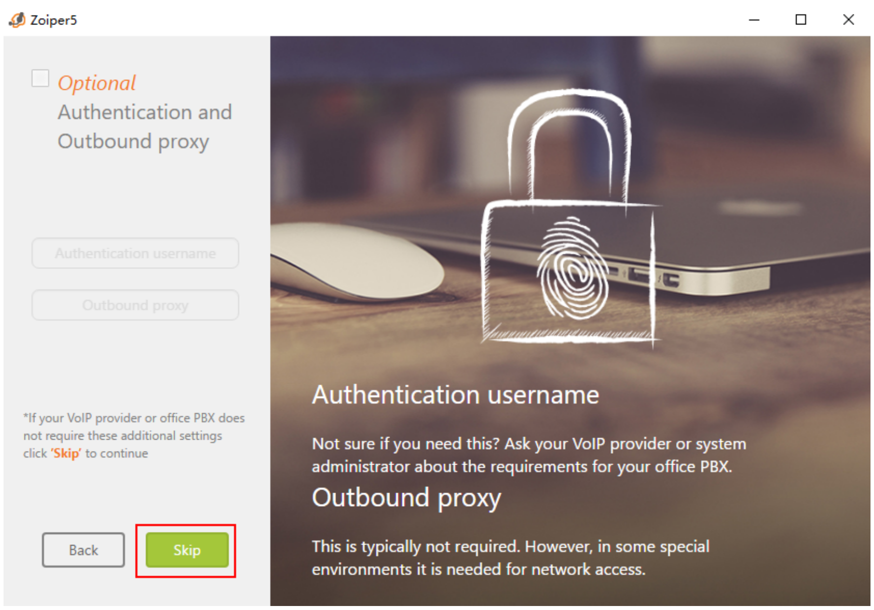
1. Launch Zoiper PC client, enter the extension number and the extension's **Registration Password**, then click **Login**.



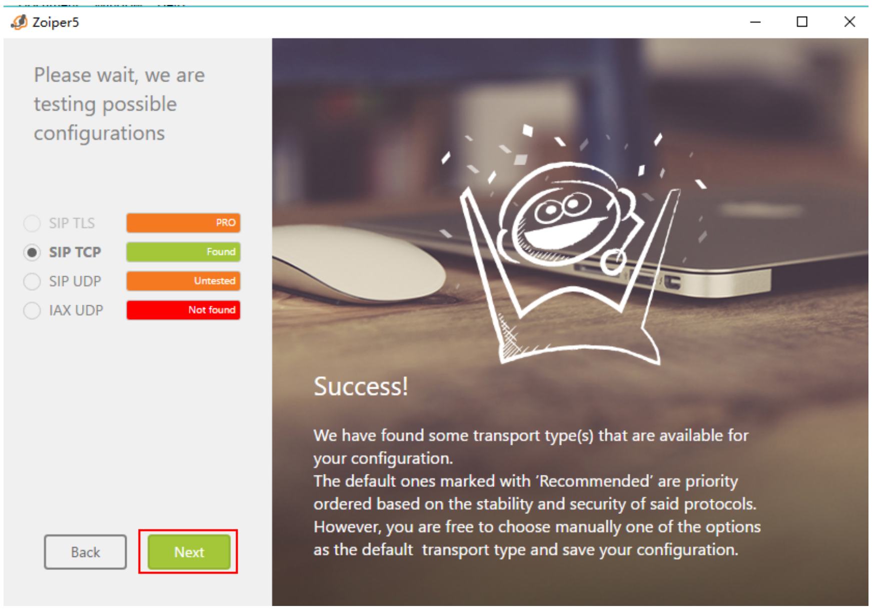
2. Enter the domain or IP address of your PBX, click **Next**.



3. Click **Skip**.



4. Click **Next**.



5. Check the account status.

If the extension is registered, you can see the status shows as the following figure.