

Grandstream Networks, Inc.

IPVideoTalk Account Configuration on 3rd Party Device



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OVERVIEW

Users could configure the **IPVideoTalk IDs** using either **SIP** or **H.323** Protocol on most popular brands of devices such as Polycom, Huawei, Yealink, and Cisco and so on.

Configuration Steps:

1. To configure the **IPVideoTalk IDs** as **SIP** or **H.323** accounts, users need to configure the options below:
 - **SIP/H.323 Account:** Users need to configure the IPVideoTalk IDs as the SIP/H.323 accounts.
 - **SIP/H.323 Password:** Users need to configure the password of the IPVideoTalk ID in the device.
 - **Server Address/Gatekeeper:** Users need to configure the IPVT10 server address in the SIP Server Address option or as H.323 Gatekeeper.
2. Other SIP Transport/Port Configuration:
 - If the SIP server port is customized port, users need to configure the server address with the customized port such as "IPVT10 Server Address: Port Number" TLS mode is recommended as the SIP Transport.
 - SIP Registration: enabled.
 - The SIP port configuration is **5060** (TCP, UDP) / **5061** (TLS), and if users want to configure the customized port for IPVT10, please configure the customized SIP port for this option.
3. Join into IPVideoTalk Meetings:
 - On the dialing interface of the device, users could input the meeting ID to join into the IPVideoTalk meeting.
 - For some certain devices (Polycom and Cisco), users need to input "Meeting ID@IP Address: Port Number" to join into the IPVideoTalk meeting.

Here are the configuration instructions for some typical devices (Some devices require special configurations):

- Polycom Real Presence Debut TM
- Huawei TEX0
- Yealink VC400
- Cisco SX20

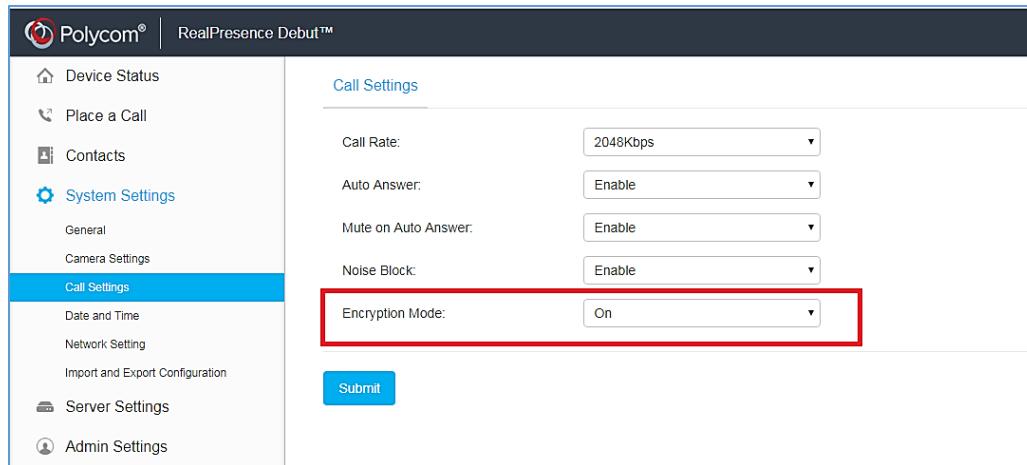


CONFIGURATION ON TYPICAL DEVICES

Configure Polycom Real Presence Debut TM

Call Settings Configuration

In order to ensure the security of the call, it is recommended to enable “Encryption Mode” in the device. This mode will force the device to use TLS protocol as the SIP Transport, otherwise, the service cannot be used normally.



The screenshot shows the Polycom RealPresence Debut Web UI. The left sidebar contains a menu with options: Device Status, Place a Call, Contacts, System Settings (highlighted), General, Camera Settings, Call Settings (highlighted), Date and Time, Network Setting, Import and Export Configuration, Server Settings, and Admin Settings. The main content area is titled 'Call Settings' and contains the following configuration options:

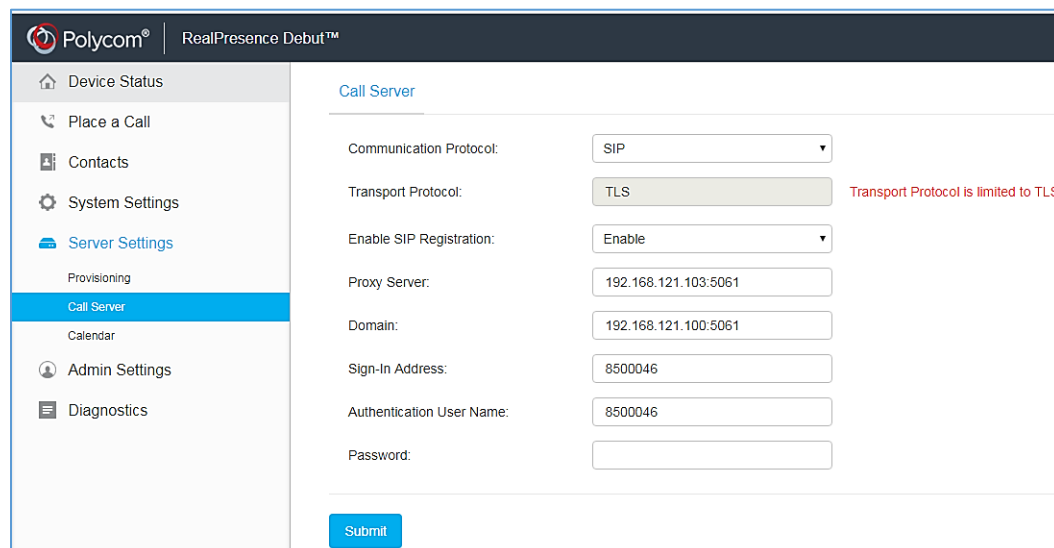
- Call Rate: 2048Kbps
- Auto Answer: Enable
- Mute on Auto Answer: Enable
- Noise Block: Enable
- Encryption Mode: On (highlighted with a red box)

A 'Submit' button is located at the bottom of the configuration area.

Figure 1: Polycom RealPresence Web UI → System Settings → Call Settings

Configure SIP Account

Users need to configure the SIP account, password, server address (Users need to fill in the port number such as “IP:Port”), and SIP protocol (Set as TLS) in the device.



The screenshot shows the Polycom RealPresence Debut Web UI. The left sidebar contains a menu with options: Device Status, Place a Call, Contacts, System Settings, Server Settings (highlighted), Provisioning, Call Server (highlighted), Calendar, Admin Settings, and Diagnostics. The main content area is titled 'Call Server' and contains the following configuration options:

- Communication Protocol: SIP
- Transport Protocol: TLS (with a red warning message: "Transport Protocol is limited to TLS")
- Enable SIP Registration: Enable
- Proxy Server: 192.168.121.103:5061
- Domain: 192.168.121.100:5061
- Sign-In Address: 8500046
- Authentication User Name: 8500046
- Password: (empty field)

A 'Submit' button is located at the bottom of the configuration area.

Figure 2: Polycom RealPresence Web UI → Server Settings → Call Server



Configure H.323 Account

Users need to go through Web Interface → **Server Settings** → **Call Server**, and select H.323 as the Communication Protocol, then select "Enable H.323 Registration" option, then configure the IPVT10 IP Address into Gatekeeper Address and he needs to specify the H.323 Name which is the IPVideoTalk ID.

Dialing Operation

Users could input the "Meeting ID@IP:Port Number" to dial into the IPVideoTalk meeting.

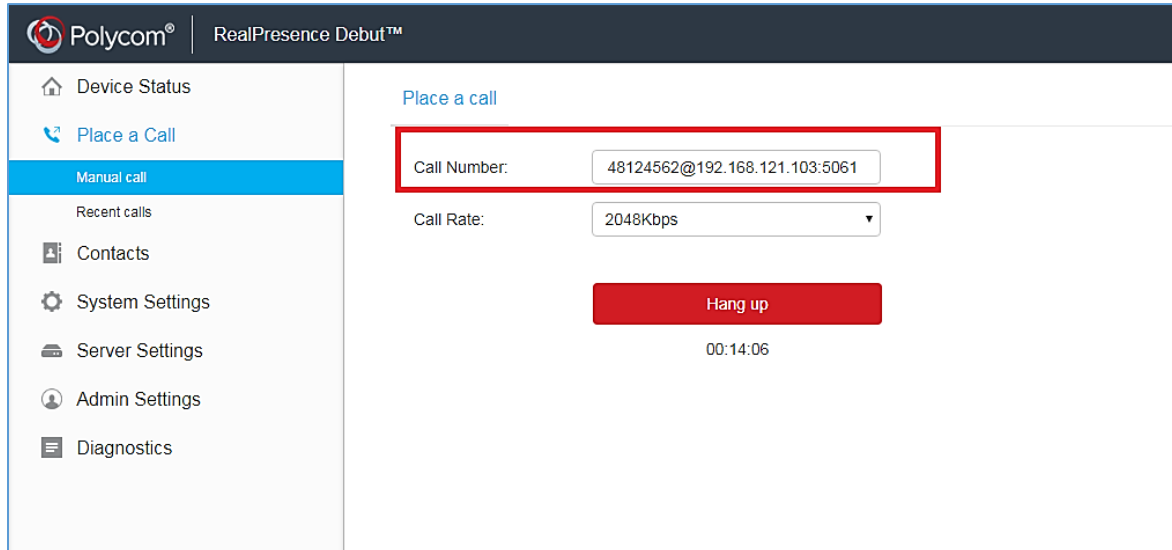


Figure 3: Polycom RealPresence Web UI → Place a Call → Manual Call

Configure Huawei TEX0

Configure SIP Account

Users need to configure the SIP account, password, server address, SIP protocol, and port number in the device.



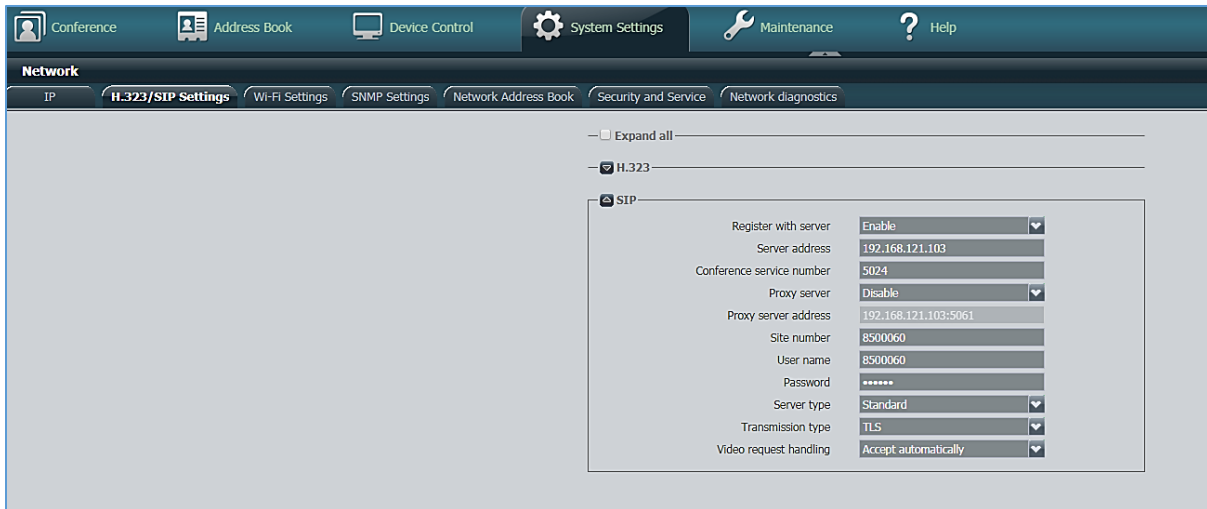


Figure 4: Huawei TEX0 → System Settings → Network → H.323/SIP Settings → SIP

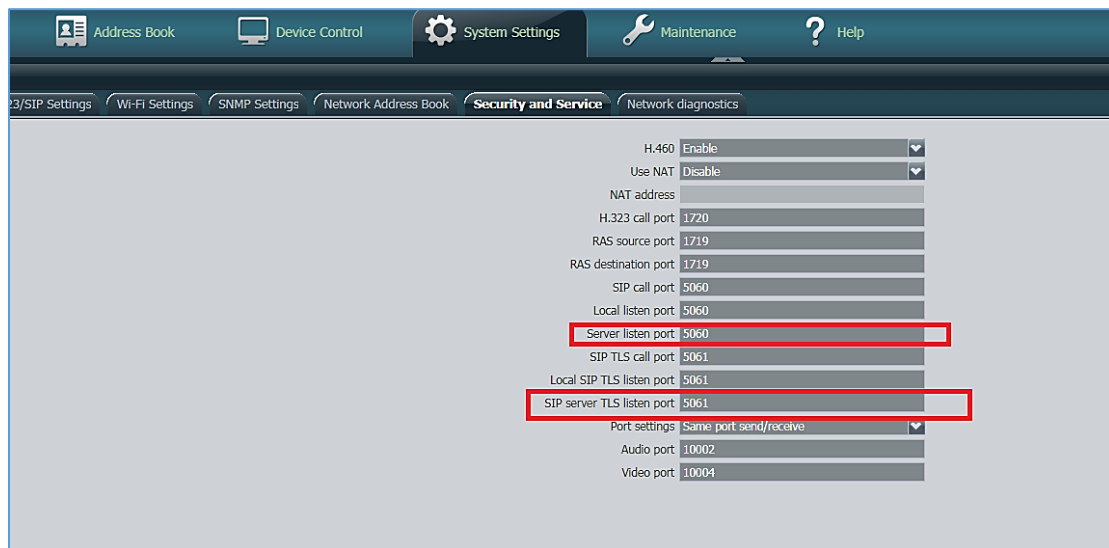


Figure 5: Huawei TEX0 → System Settings → Network → Security and Service

Configure SRTP

Users have to enable “Encryption” option in the device before using the device, otherwise, it will cause the called function abnormal issues.



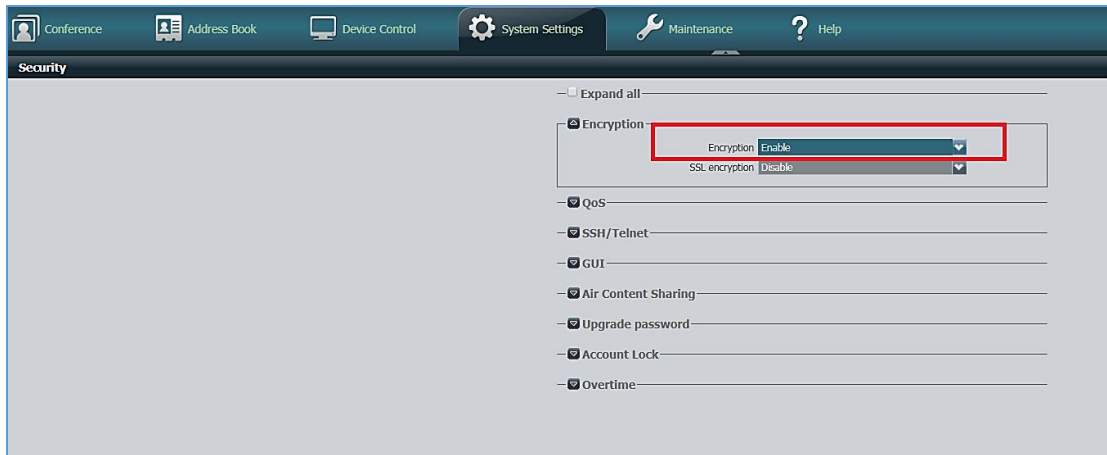


Figure 6: Huawei TEX0 → System Settings → Security

Configure H.323 Account

Users need to select Enable GK, Set GK Registration Mode to Manual, enter the IPVT10 IP address into GK Address and enter IPVideoTalk ID in Authentication User name and the IPVideoTalk Password into Password.

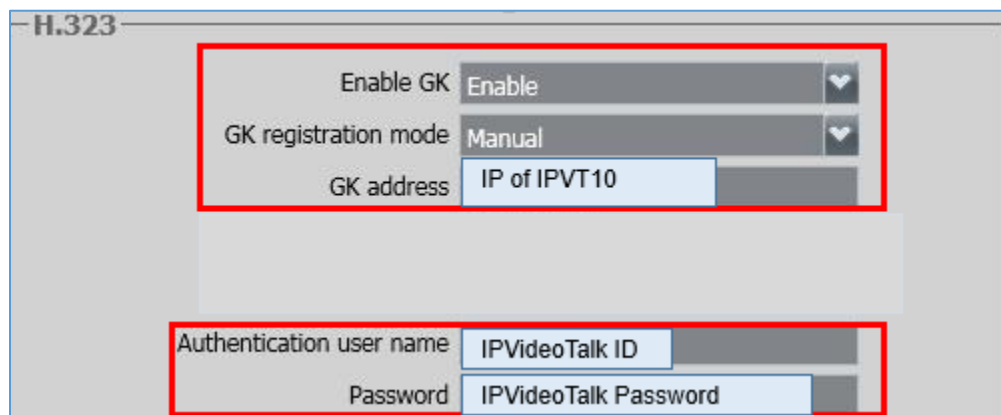


Figure 7: Huawei TEX0 → System Settings → Network → H.323/SIP Settings → H.323

Dialing Operation

Users could input the IPVideoTalk ID on the dialing interface to dial into the IPVideoTalk meeting. Please note that users need to select option “Line Type” as “SIP”.



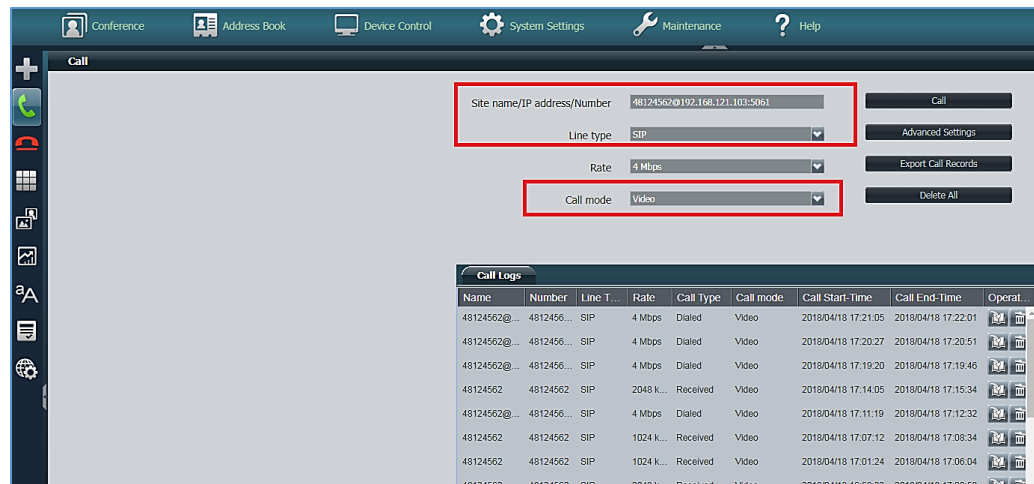


Figure 8: Huawei TEXO → Dialing Page

Configure Yealink VC400

Configure SIP Account

Users need to configure the SIP account, password, server address, SIP protocol, and port number in the device. Users also need to configure “SRTP” option as “Compulsory”, “DTMF Type” option as “RFC2833” before using the device for IPVideoTalk services.

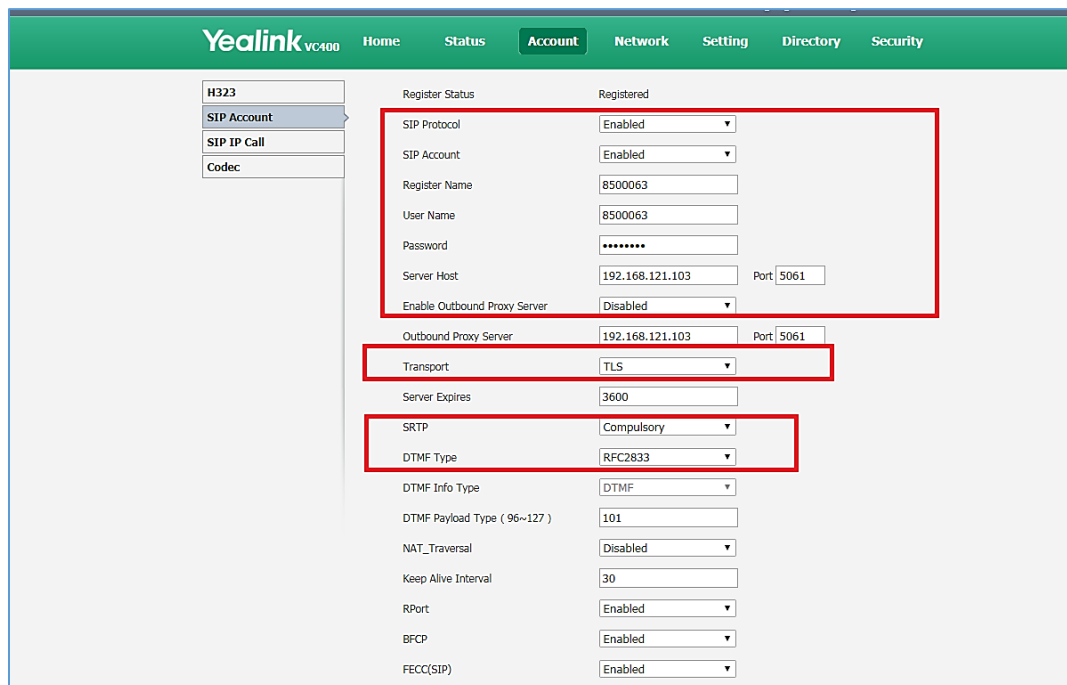


Figure 9: Yealink VC400 → Account → SIP Account

Configure TLS

When users set the “SIP Transport” as “TLS” in Yealink VC400, users need to disable option “Only Accept Trusted Certificates”, otherwise, the TLS connection will be failed.



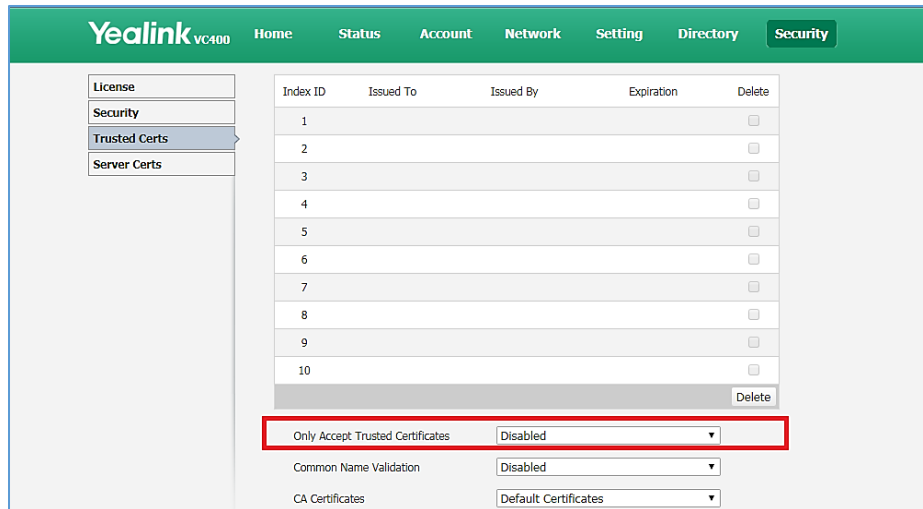


Figure 10: Yealink VC400 → Security → Trusted Certs

Configure H.323 Account

The following configurations are necessary:

- H.323 Protocol: Choose Enabled.
- H.323 Account: Choose Enabled.
- Gatekeeper Mode: Manual
- Gatekeeper IP Address: Configures the IP address of the IPVT10
- Gatekeeper Authentication Mode: Choose "Enabled" to enable authentication.
- Gatekeeper Username: The username is the IPVideoTalk ID
- Gatekeeper Password: The password of the IPVideoTalk ID

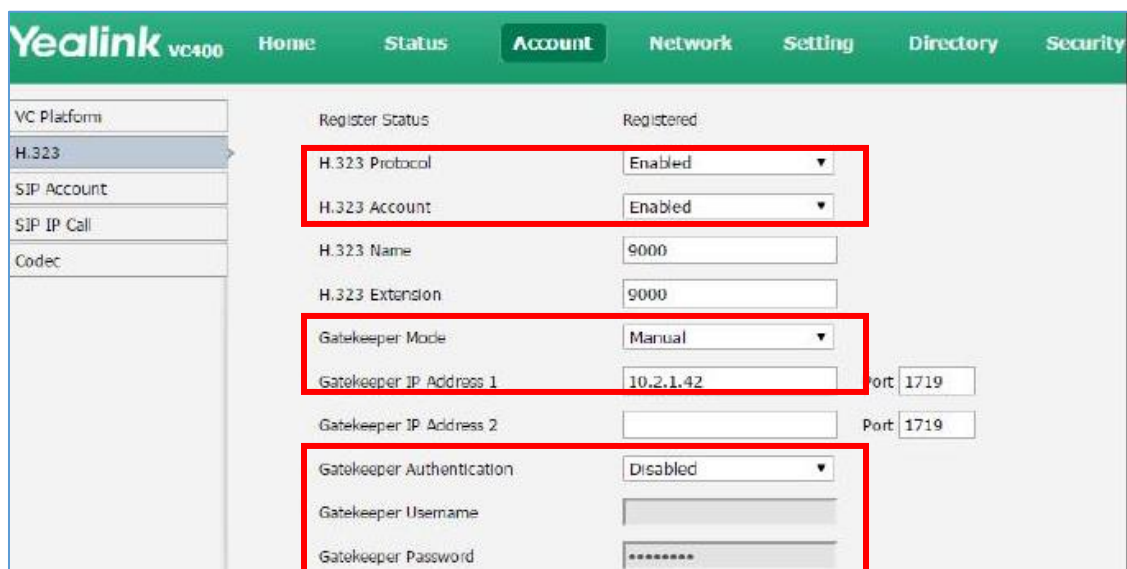


Figure 11: Yealink VC400 → H.323



Dialing Operation

Users could input the IPVideoTalk ID on the dialing interface to dial into the IPVideoTalk meeting.

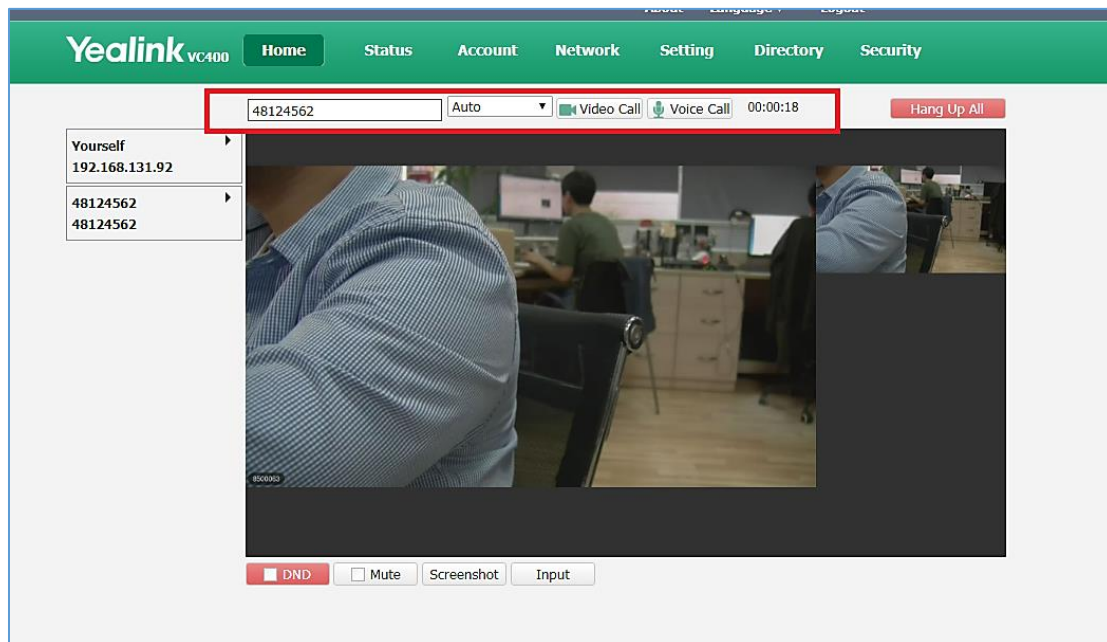


Figure 12: Yealink VC400 → Home

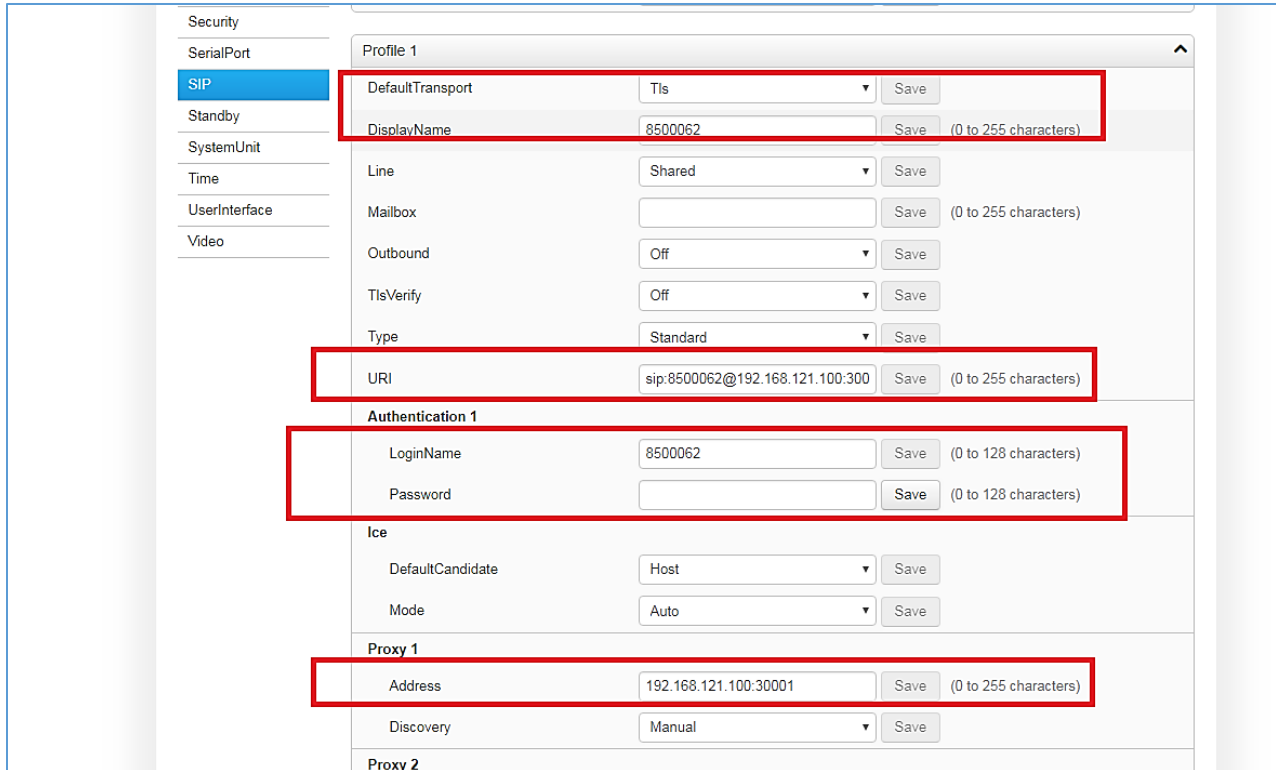
Configure Cisco SX20

Configure SIP Account

The following configurations are necessary:

- SIP Transport: TLS ONLY
- URL: IPVideoTalk ID@IPVT10 Server Address:Port Number
- Login Name: IPVideoTalk ID
- Password: The password of the IPVideoTalk ID
- Address: IPVT10 Server Address:Port Number





The screenshot shows the 'SIP' configuration page for 'Profile 1' on a Cisco SX20 device. The left sidebar lists various configuration categories: Security, SerialPort, SIP (selected), Standby, SystemUnit, Time, UserInterface, and Video. The main content area is divided into sections: Profile 1, Authentication 1, Ice, and Proxy 1. Red boxes highlight the following fields:

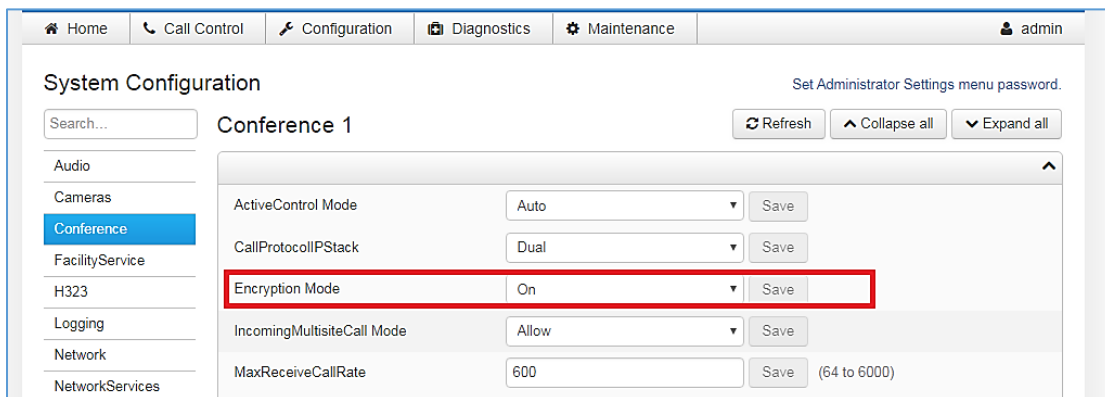
- Profile 1:** DefaultTransport (Tls), DisplayName (8500062), URI (sip:8500062@192.168.121.100:300).
- Authentication 1:** LoginName (8500062), Password (empty).
- Proxy 1:** Address (192.168.121.100:30001).

Other visible fields include Line (Shared), Mailbox, Outbound (Off), TlsVerify (Off), Type (Standard), DefaultCandidate (Host), Mode (Auto), and Discovery (Manual).

Figure 13: Cisco SX20 → Configuration → SIP

Configure SRTP

In order to ensure the security of the call, it is recommended to enable “Encryption Mode” in the device. This mode will force the device to only use TLS protocol as the SIP Transport, otherwise, the service cannot be used normally.



The screenshot shows the 'Conference' configuration page for 'Conference 1' on a Cisco SX20 device. The left sidebar lists various configuration categories: Home, Call Control, Configuration (selected), Diagnostics, Maintenance, and admin. The main content area is divided into sections: System Configuration, Conference 1, and a search bar. Red boxes highlight the following fields:

- Conference 1:** Encryption Mode (On).

Other visible fields include ActiveControl Mode (Auto), CallProtocolIPStack (Dual), IncomingMultisiteCall Mode (Allow), and MaxReceiveCallRate (600).

Figure 14: Cisco SX20 → Configuration → Conference

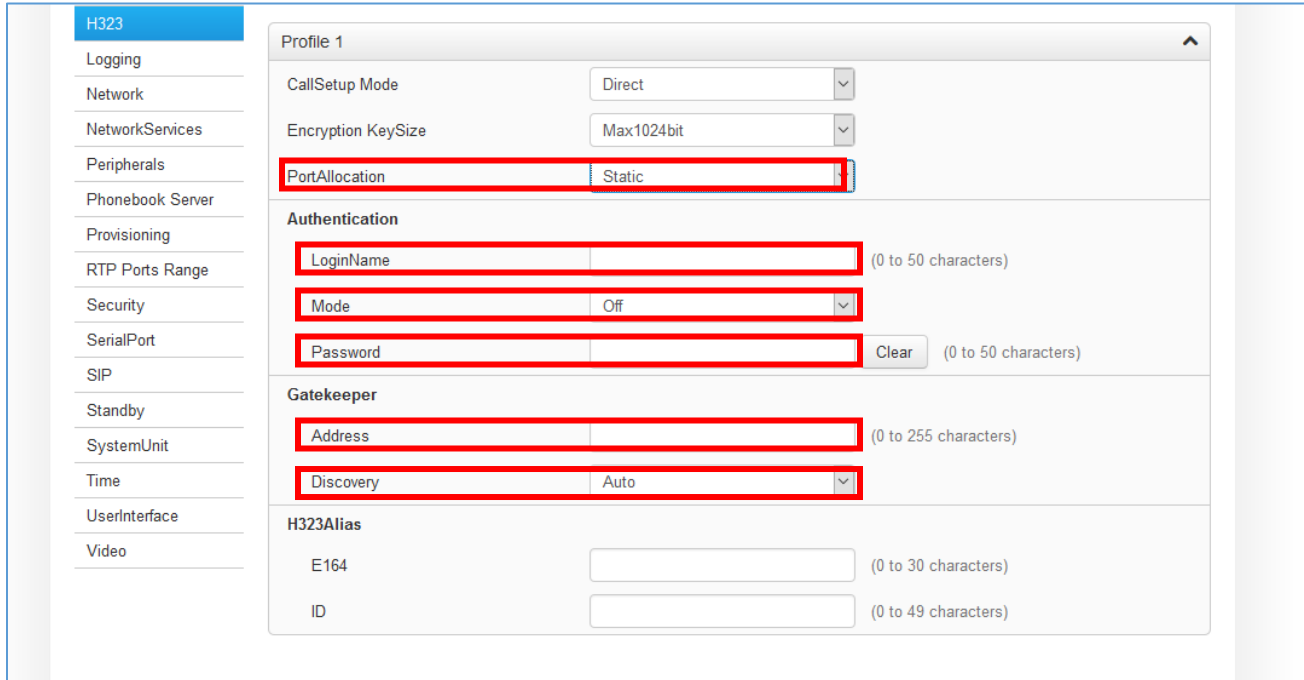
Configure H.323 Account

The following configurations are necessary:

- PortAllocation: Choose Dynamic
- LoginName: IPVideoTalk ID



- Mode: Choose On to enable authentication.
- Password: The password of the IPVideoTalk ID
- Address: IPVT10 Server Address
- Discovery: Manual.



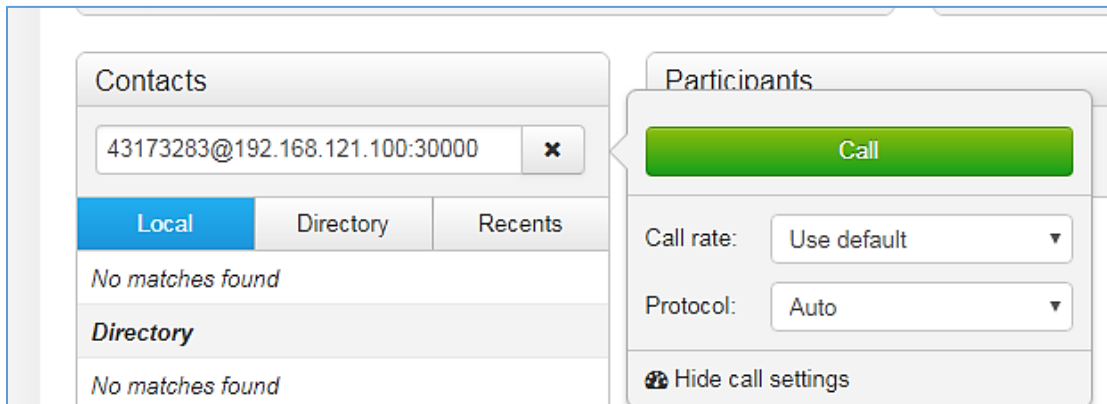
The screenshot shows the H.323 configuration page for Profile 1. The left sidebar lists various configuration categories, with H.323 selected. The main content area contains the following settings:

- CallSetup Mode:** Direct
- Encryption KeySize:** Max1024bit
- PortAllocation:** Static
- Authentication:**
 - LoginName:** (0 to 50 characters)
 - Mode:** Off
 - Password:** (0 to 50 characters) with a Clear button.
- Gatekeeper:**
 - Address:** (0 to 255 characters)
 - Discovery:** Auto
- H323Alias:**
 - E164:** (0 to 30 characters)
 - ID:** (0 to 49 characters)

Figure 15: Cisco SX20 → Configuration → H.323

Dialing Operation

Users could input the “IPVideoTak Meeting ID@IP:Port Number” on the dialing interface to join into the IPVideoTalk meeting.



The screenshot shows the Dial Page interface. It is divided into two main sections: Contacts and Participants.

- Contacts:**
 - A search bar contains the text: 43173283@192.168.121.100:30000.
 - Below the search bar are tabs for Local, Directory, and Recents.
 - Under the Local tab, it says "No matches found".
 - Under the Directory tab, it says "No matches found".
- Participants:**
 - A large green **Call** button is at the top.
 - Below the button are settings:
 - Call rate:** Use default
 - Protocol:** Auto
 - At the bottom, there is a link to **Hide call settings**.

Figure 16: Cisco SX20 → Dial Page

