



Grandstream Networks, Inc.

IPVideoTalk Account Configuration on 3rd Party Device



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OVERVIEW

Users could configure the IPVideoTalk IDs on most popular brands of devices such as Polycom, Huawei, Yealink, Cisco and so on.

Configuration Steps:

1. To configure the IPVideoTalk IDs as SIP accounts, users need to configure the options below:
 - SIP Account: Users need to configure the IPVideoTalk IDs as the SIP accounts.
 - SIP Password: Users need to configure the password of the IPVideoTalk ID in the device.
 - Server Address: Users need to configure the IPVT10 server address in the SIP Server Address option. 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”.
 - SIP Registration: Enabled
2. SIP Transport/Port Configuration:
 - TLS mode is recommended as the SIP Transport.
 - 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):

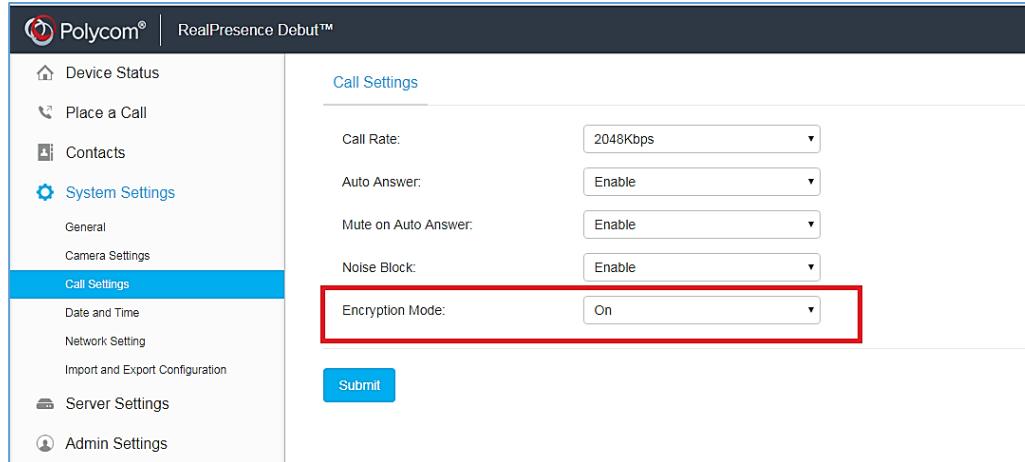
- Polycom Real Presence Debut TM
- Huawei TEX0
- Yealink VC400
- Cisco SX20

CONFIGURATION ON TYPICAL DEVICES

Configure Polycom Real Presence Debut TM

Dialing 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 interface. The left sidebar has links for Device Status, Place a Call, Contacts, System Settings (which is selected), Call Settings (selected), Date and Time, Network Setting, Import and Export Configuration, Server Settings, and Admin Settings. The main panel is titled "Call Settings" and contains the following fields:

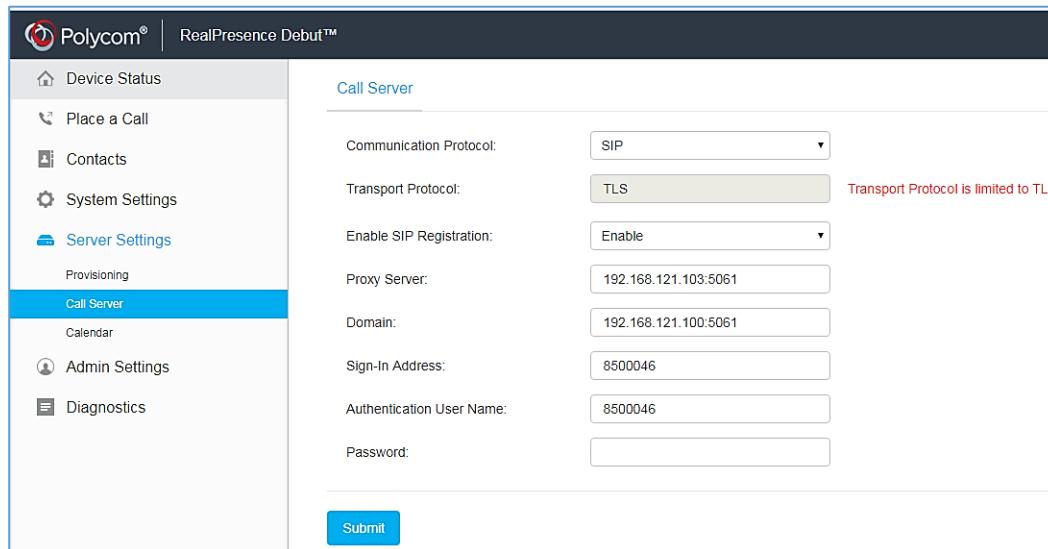
- Call Rate: 2048Kbps
- Auto Answer: Enable
- Mute on Auto Answer: Enable
- Noise Block: Enable
- Encryption Mode: On (highlighted with a red box)

A "Submit" button is at the bottom.

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 interface. The left sidebar has links for Device Status, Place a Call, Contacts, System Settings, Server Settings (selected), Provisioning, Call Server (selected), Calendar, Admin Settings, and Diagnostics. The main panel is titled "Call Server" and contains the following fields:

- Communication Protocol: SIP
- Transport Protocol: TLS (note: 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 at the bottom.

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



Dialing Operation

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

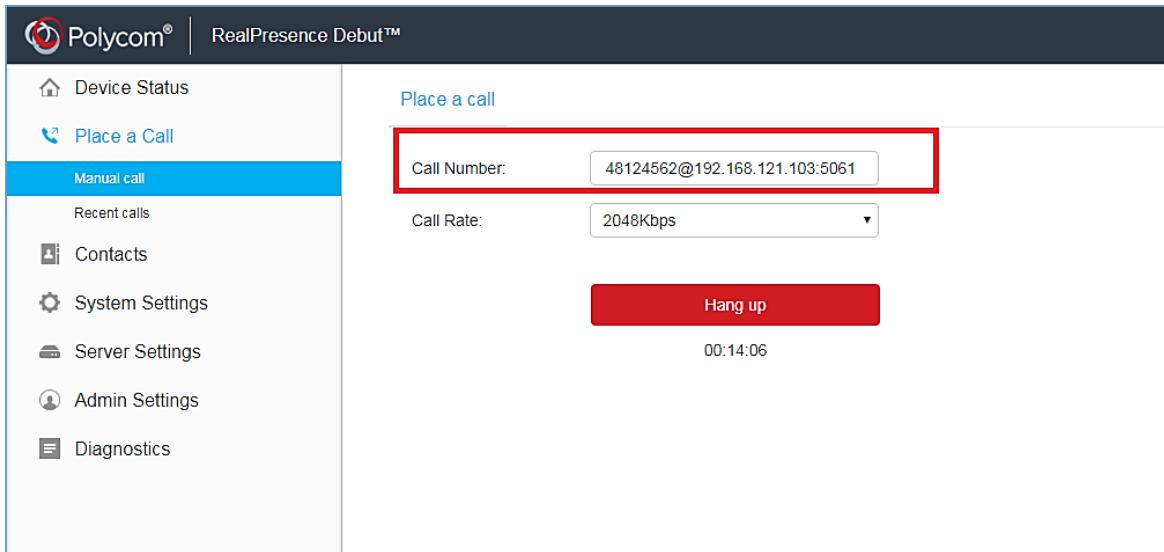


Figure 3: Polycom RealPresence Web UI → Place a Call → Manual Call

Configure Huawei TEXO

Configure SIP Account

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

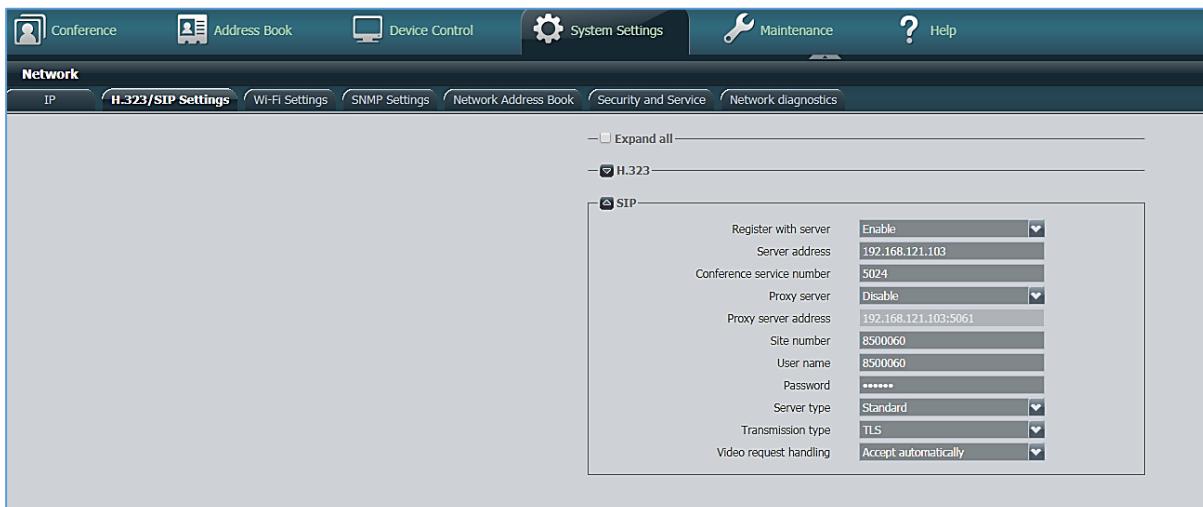


Figure 4: Huawei TEXO → System Settings → Network → H.323/SIP Settings → SIP



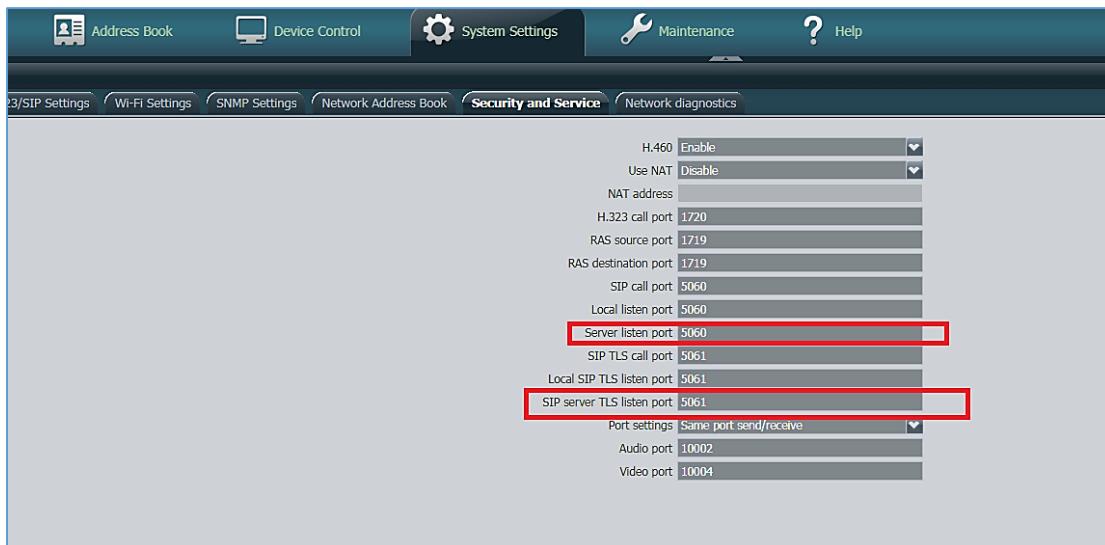


Figure 5: Huawei TEX0 → System Settings → Network → Security and Service

Configure SRTP

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

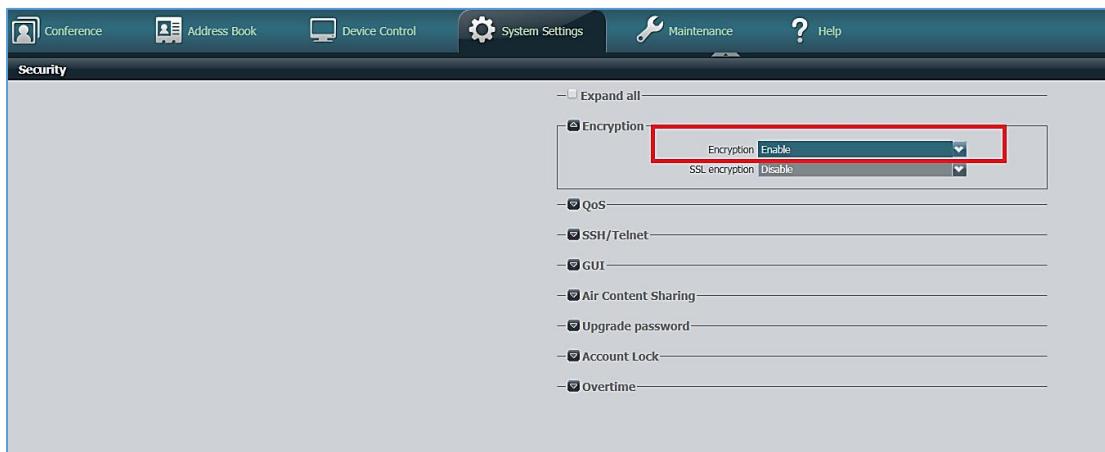


Figure 6: Huawei TEX0 → System Settings → Security

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”.



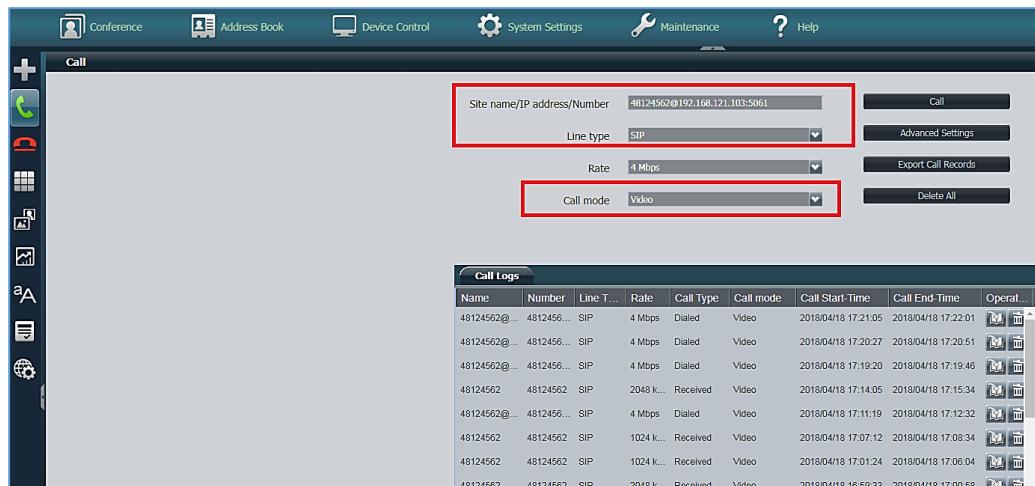


Figure 7: Huawei TEX0 → Dialing Page

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.

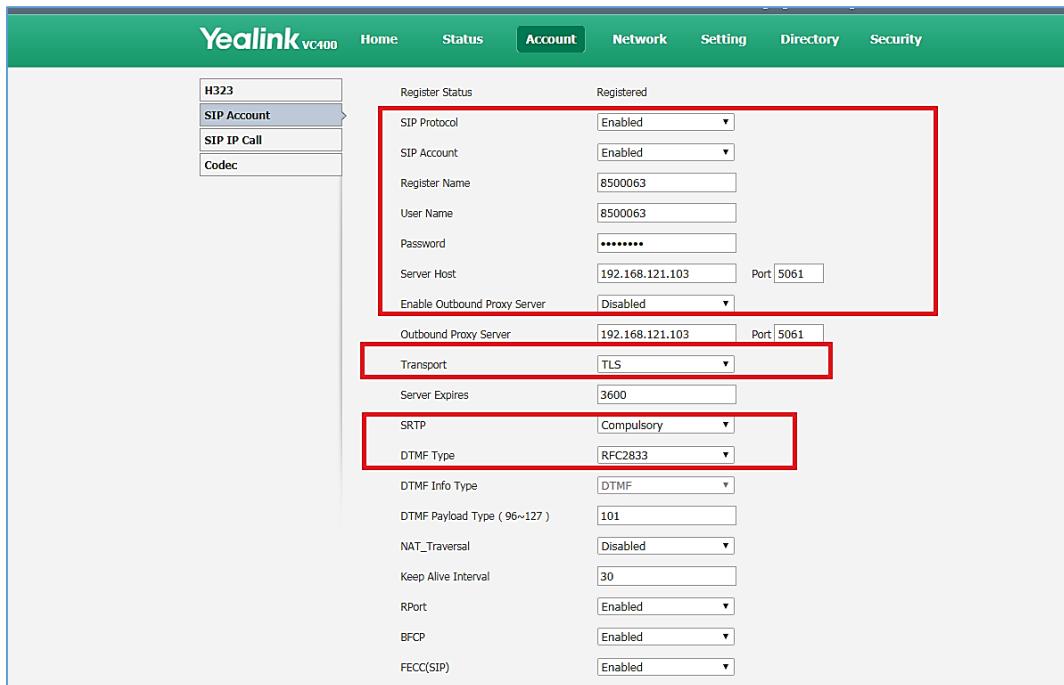


Figure 8: Yealink VC400 → Account → SIP Account

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.



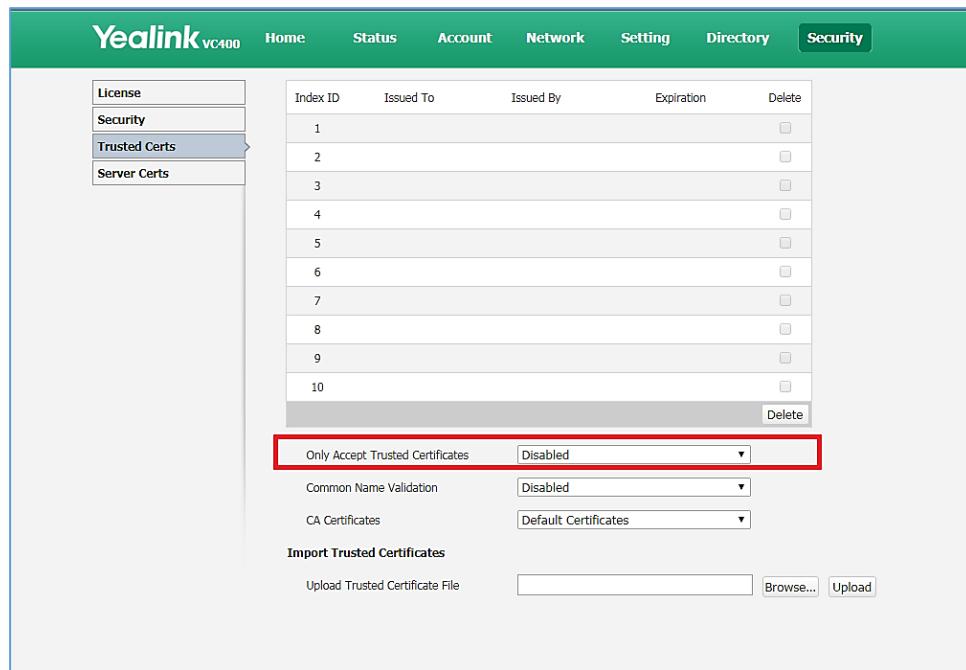


Figure 9: Yealink VC400 → Security → Trusted Certs

Dialing Operation

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

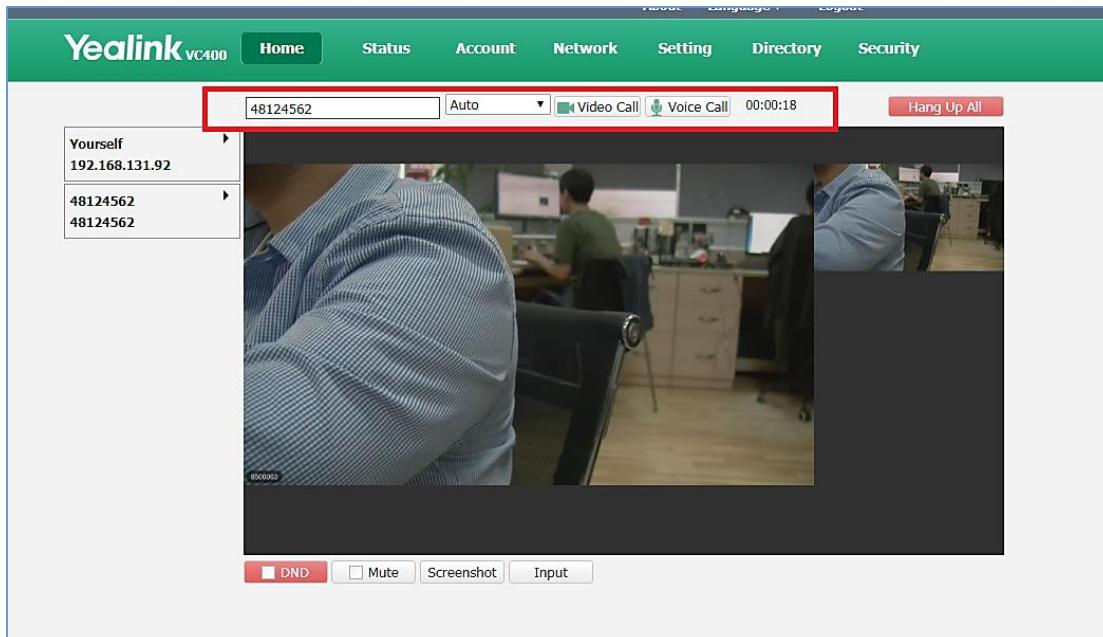


Figure 10: Yealink VC400 → Home

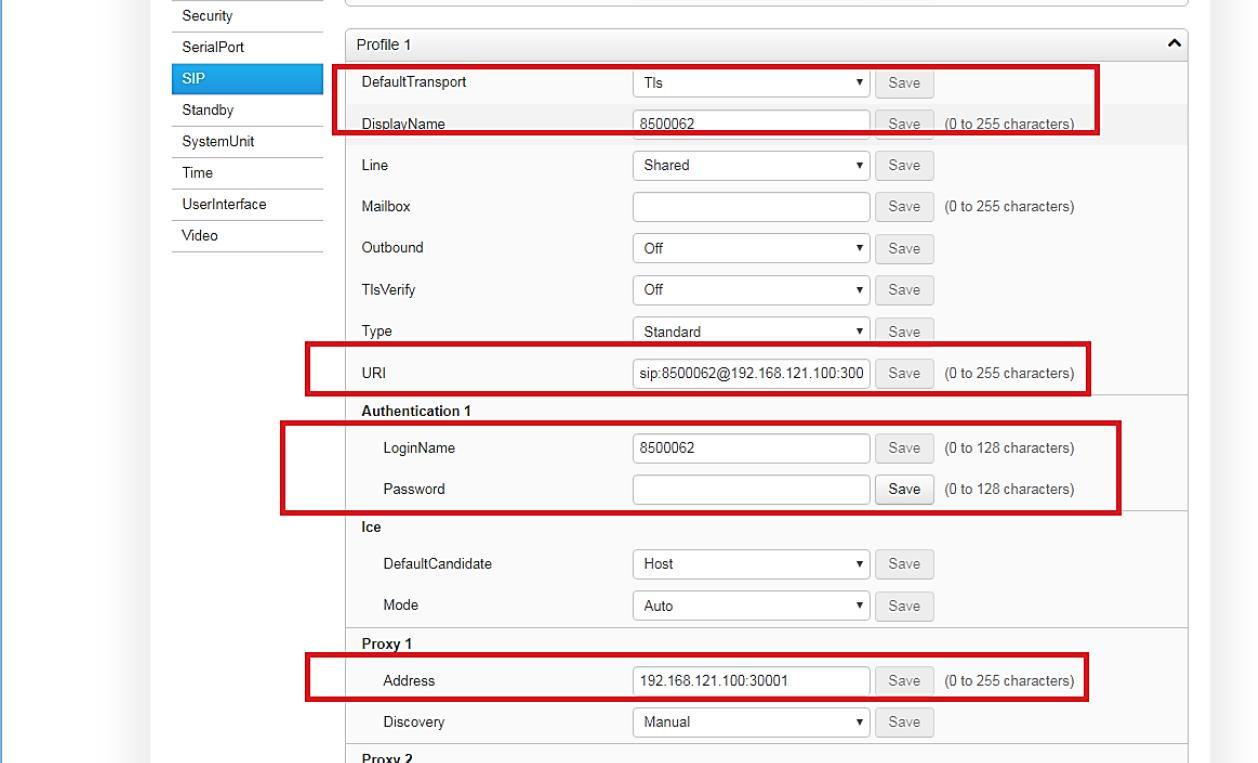


Configure Cisco SX20

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 Cisco SX20 configuration interface for SIP. The left sidebar lists categories: Security, SerialPort, SIP (selected), Standby, SystemUnit, Time, UserInterface, and Video. The main area is titled 'Profile 1' and contains the following fields:

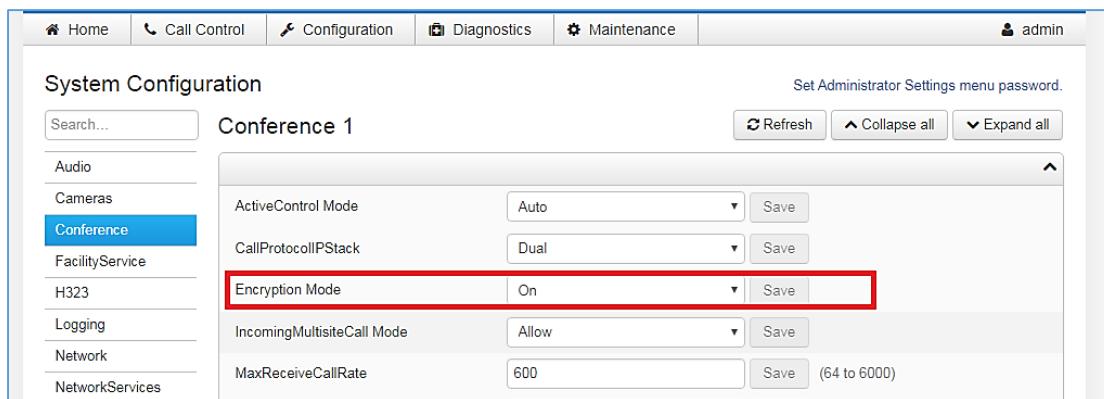
- DefaultTransport:** Tls (highlighted with a red box)
- DisplayName:** 8500062 (highlighted with a red box)
- Line:** Shared
- Mailbox:** (empty)
- Outbound:** Off
- TlsVerify:** Off
- Type:** Standard
- URI:** sip:8500062@192.168.121.100:300 (highlighted with a red box)
- Authentication 1** section:
 - LoginName:** 8500062
 - Password:** (empty)
- Ice** section:
 - DefaultCandidate:** Host
 - Mode:** Auto
- Proxy 1** section:
 - Address:** 192.168.121.100:30001 (highlighted with a red box)
 - Discovery:** Manual
- Proxy 2** section: (empty)

Figure 11: 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.



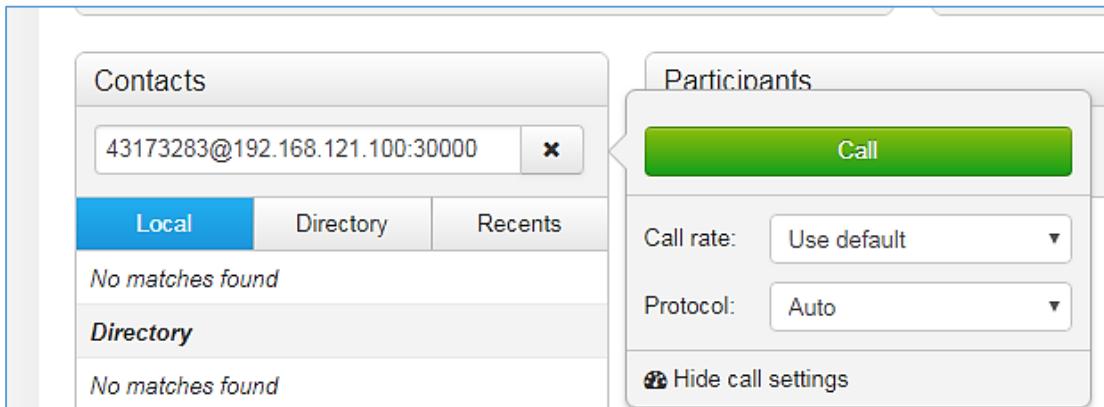


The screenshot shows the Grandstream GXV3240 web interface under the 'System Configuration' section, specifically the 'Conference' tab. On the left sidebar, 'Conference' is selected. The main area displays configuration settings for 'Conference 1'. One setting, 'Encryption Mode', has its dropdown menu open with 'On' selected, and this entire row is highlighted with a red box. Other settings shown include 'ActiveControl Mode' (Auto), 'CallProtocol/IPStack' (Dual), 'IncomingMultisiteCall Mode' (Allow), and 'MaxReceiveCallRate' (600). Buttons for 'Save' and 'Cancel' are visible next to each setting.

Figure 12: Cisco SX20 → Configuration → Conference

Dialing Operation

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



The screenshot shows the Cisco SX20 dial page. On the left, there's a 'Contacts' section containing an entry '43173283@192.168.121.100:30000'. Below it are tabs for 'Local', 'Directory', and 'Recents', with 'Local' being the active tab. A message 'No matches found' is displayed under the directory tab. On the right, there's a 'Participants' section with a large green 'Call' button. Below the button are dropdown menus for 'Call rate' (set to 'Use default') and 'Protocol' (set to 'Auto'). There's also a link 'Hide call settings'.

Figure 13: Cisco SX20 → Dial Page

