

# Grandstream Networks, Inc.

# IPVideoTalk Account Configuration on 3rd Party Device





## **Table of Contents**

OVERVIEW	. 4
CONFIGURATION ON TYPICAL DEVICES	5
Configure Polycom Real Presence Debut TM	5
Dialing Configuration	5
Configure SIP Account	5
Dialing Operation	6
Configure Huawei TEX0	6
Configure SIP Account	6
Configure SRTP	7
Dialing Operation	7
Configure Yealink VC400	8
Configure SIP Account	8
Configure TLS	8
Dialing Operation	9
Configure Cisco SX20	10
Configure SIP Account	.10
Configure SRTP	.10
Dialing Operation	. 11





# Table of figures

Figure 1: Polycom RealPresence Web UI $\rightarrow$ System Settings $\rightarrow$ Call Settings	5
Figure 2: Polycom RealPresence Web UI $\rightarrow$ Server Settings $\rightarrow$ Call Server	5
Figure 3: Polycom RealPresence Web UI $ ightarrow$ Place a Call $ ightarrow$ Manual Call	6
Figure 4: Huawei TEX0 $\rightarrow$ System Settings $\rightarrow$ Network $\rightarrow$ H.323/SIP Settings $\rightarrow$ SIP	6
Figure 5: Huawei TEX0 $\rightarrow$ System Settings $\rightarrow$ Network $\rightarrow$ Security and Service	7
Figure 6: Huawei TEX0 $\rightarrow$ System Settings $\rightarrow$ Security	7
Figure 7: Huawei TEX0 $\rightarrow$ Dialing Page	8
Figure 8: Yealink VC400 $\rightarrow$ Account $\rightarrow$ SIP Account	8
Figure 9: Yealink VC400 $\rightarrow$ Security $\rightarrow$ Trusted Certs	9
Figure 10: Yealink VC400 $\rightarrow$ Home	9
Figure 11: Cisco SX20 $\rightarrow$ Configuration $\rightarrow$ SIP	. 10
Figure 12: Cisco SX20 $\rightarrow$ Configuration $\rightarrow$ Conference	. 11
Figure 13: Cisco SX20 $\rightarrow$ Dial Page	. 11



### **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) / **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.

	Debut™		
Device Status	Call Settings		
Place a Call			
Contacts	Call Rate:	2048Kbps	T
System Settings	Auto Answer:	Enable	T
General	Mute on Auto Answer:	Enable	¥
Camera Settings	Noise Block:	Enable	•
Call Settings			
Date and Time	Encryption Mode:	On	<b>T</b>
Network Setting			
Import and Export Configuration	Ruhmit		
Server Settings	Subrin		
Admin Settings			

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.

Polycom <sup>®</sup> RealPresence	Pebut™		
Device Status	Call Server		
Place a Call			
Contacts	Communication Protocol:	SIP	T
System Settings	Transport Protocol:	TLS	Transport Protocol is limited to TLS
📾 Server Settings	Enable SIP Registration:	Enable	T
Provisioning	Proxy Server:	192.168.121.103:5061	
Call Server	Domain	102 168 121 100 5061	
Calendar	Domain.	192.100.121.100.0001	
Admin Settings	Sign-In Address:	8500046	
Diagnostics	Authentication User Name:	8500046	
	Password:		
	Submit		
Figure 2: P	olycom RealPresence Web	UI → Server Settings →	Call Server





#### **Dialing Operation**

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

	ebut™		
☆ Device Status	Place a call		
V Place a Call			
Manual call	Call Number:	48124562@192.168.121.103:5061	
Recent calls	Call Rate:	2048Kbps •	
Contacts			
System Settings		Hang up	
Server Settings		00:14:06	
Admin Settings			
Diagnostics			
Figure 3: Po	olvcom RealPresence	e 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.

Conferen	ice Address Book	Device Control	System Settings	? Help
Network				
IP	H.323/SIP Settings Wi-Fi Settings	SNMP Settings Network Address	s Book Security and Service Network diagnostics	
			- D Expand all	
			<b>Q</b> U 222	
			- MI1.525-	
			Register with server	Enable
			Server address	192.168.121.103
			Conference service number	5024
			Proxy server	Disable
			Proxy server address	192.168.121.103:5061
			Site number	8500060
			User name	8500060
			Password	
			Server type	Standard 💌
			Transmission type	TLS 👻
			Video request handling	Accept automatically
			<u></u>	

Figure 4: Huawei TEX0  $\rightarrow$  System Settings  $\rightarrow$  Network  $\rightarrow$  H.323/SIP Settings  $\rightarrow$  SIP





Address Book	evice Control System Settings	Maintenance	? Help	
3/SIP Settings Wi-Fi Settings SNMP Settir	igs Network Address Book Security and Ser	vice Network diagnostics		
		H.460 Enable	<b>~</b>	
		Use NAT Disable NAT address	~	
		H.323 call port 1720		
	R	AS destination port 1719		
		SIP call port 5060		
	1	Server listen port 5060		
	Local	SIP TLS call port 5061 SIP TLS listen port 5061		_
	SIP se	erver TLS listen port 5061 Port settings Same port send/rec	eive 🗸	]
		Audio port 10002		
		Video port 10004		

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.

Conference	Address Book	Device Control	System Settings	Haintenance	? Help		
Security							
			— 🗆 Ехра	nd all			
			- 🖾 Encr	yption - Encryption SSL encryption	Enable Disable	V V	1
			- 🖾 QoS-				
			- 🖬 SSH/	/Telnet			
			- 🖾 GUI -	<u></u>			
			— 🖾 Air C	ontent Sharing			
			— 🖬 Upgr	ade password			
			- 🖾 Acco	unt Lock			
			— 🖾 Over	time			

Figure 6: Huawei TEX0  $\rightarrow$  System Settings  $\rightarrow$  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".





	Conference	Address Book	Device Control	🗘 sy	stem Settin	ıgs	ا مکن	laintenance	?	Help		
+	Call	_	_			_	_			_	_	
C				Site name/	IP address/	(Number	4812456	2@192.168.121	1.103:5061		Call	
					L	ine type	SIP			~	Advanced Setting	S
						Rate	4 Mbps			~	Export Call Recor	is
					C	all mode	Video			~	Delete All	
Ē												
				Colling	_							
aA				Name	Number	Line T	Rate	Call Type	Call mode	Call Start-Tim	ne Call End-Time	Operat
目				48124562@	4812456	SIP	4 Mbps	Dialed	Video	2018/04/18 17:2	21:05 2018/04/18 17:22:	и 🕅 🖬 🕯
				48124562@	4812456	SIP	4 Mbps	Dialed	Video	2018/04/18 17:2	20:27 2018/04/18 17:20:	
<i>e</i> o I				48124562	48124562	SIP	2048 k	Received	Video	2018/04/18 17:	14:05 2018/04/18 17:15:	4 M T
l				48124562@	4812456	SIP	4 Mbps	Dialed	Video	2018/04/18 17:*	11:19 2018/04/18 17:12:	12 1 1
				48124562	48124562	SIP	1024 k	Received	Video	2018/04/18 17:0	07:12 2018/04/18 17:08:	94 💓 🛅
				48124562	48124562	SIP	1024 k	Received	Video	2018/04/18 17:0	01:24 2018/04/18 17:06: 59:33 2018/04/18 17:00:	4 🔛 🖬

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.

Yealink vc400	Home Status	Account Network	Setting Director	y Security
H323 SIP Account SIP IP Call Codec	Register Status SIP Protocol SIP Account Register Name User Name Password Server Host	Registered Enabled Enabled 8500063 8500063 ••••••• 192.168.121.103	v     v     v     Port 5061	
	Outbound Proxy Server	192.168.121.103	Port 5061	
	Server Expires	3600		
	SRTP DTMF Type	Compulsory RFC2833	T	
	DTMF Info Type DTMF Payload Type ( 96- NAT_Traversal Keep Alive Interval	-127 ) DTMF Disabled 30	¥ 	
	RPort BFCP FECC(SIP)	Enabled Enabled Enabled	T T	

Figure 8: Yealink VC400  $\rightarrow$  Account  $\rightarrow$  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.





Yealink vc400	Home	Status Acco	unt Network	Setting Direc	tory Secu	rity
License	Index ID	Issued To	Issued By	Expiration	Delete	
Security	1					
Trusted Certs	2					
Server Certs	3					
	4					
	5					
	6					
	7					
	8					
	9					
	10					
					Delete	
	Only Acce	ept Trusted Certificates	Disabled		•	
	Common	Name Validation	Disabled		•	
	CA Certifi	icates	Default Certif	icates	•	
	Import Trus	sted Certificates				
	Upload T	rusted Certificate File			Browse U	Jpload

Figure 9: Yealink VC400 → Security → Trusted Certs

#### **Dialing Operation**

 Year
 Year

 Year
 Market Year

 Image: Market Year
 Status
 Account
 Network
 Setting
 Directory
 Security

 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year

 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year

 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year

 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year

 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year

 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year

 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year

 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year

 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Year
 Image: Market Y

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/TCP
- 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

Secu	rity					
Seria	lPort	Profile 1				
SIP		DefaultTransport	TIs	•	Save	
Stand	dby	DisplavName	8500062		Save	(0 to 255 characters)
Syste	emUnit					
Time		Line	Shared	•	Save	
User	Interface	Mailbox			Save	(0 to 255 characters)
Video	0	Outbound	Off	•	Save	
		TIsVerify	Off	T	Save	
		Туре	Standard	•	Save	
		URI	sip:8500062@192.168.121.10	0:300	Save	(0 to 255 characters)
		Authentication 1				
		LoginName	8500062		Save	(0 to 128 characters)
		Password			Save	(0 to 128 characters)
		Ice				
		DefaultCandidate	Host	T	Save	
		Mode	Auto	•	Save	
		Proxy 1				
		Address	192.168.121.100:30001		Save	(0 to 255 characters)
		Discovery	Manual	T	Save	
		Proxy 2				

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 use TLS protocol as the SIP Transport, otherwise, the service cannot be used normally.





A Home	📞 Call Control	Configuration	Diagnostics	Maintenance				💄 admin
System C	Configuration					Set	Administrator Setting	s menu password.
Search Conference 1						C Refresh	▲ Collapse all	✓ Expand all
Audio								^
Cameras	Activ	eControl Mode	Auto		T	Save		
Conference	Call	ProtocollDStack	Dual		-	Sava		
FacilityServic	e	TOLOCOILE SLACK	Duai			Save		
H323	Encr	yption Mode	On		•	Save		
Logging	Inco	mingMultisiteCall Mode	Allow	/	T	Save		
Network	Maria	Density On IIDeta	600				(04 +- 0000)	
NetworkServ	ces	ReceiveCallRate	600			Save	(64 to 6000)	

Figure 12: Cisco SX20 → Configuration → Conference

#### **Dialing Operation**

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

Contacts			Participants			
43173283@19	2.168.121.100:30	× 0000		Call		
Local	Directory	Recents	Call rate:	Use default		
No matches four	nd		Protocol:			
Directory				Auto	•	
No matches fou	nd		▲ Hide call settings			

Figure 13: Cisco SX20 → Dial Page

