

Grandstream Networks, Inc.

IPVideoTalk Service Configuration Guide on UCM





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OVERVIEW

This document introduces the IPVideoTalk service configuration on Grandstream UCM device with IPVT10 server.

1. Connect Grandstream UCM with IPVT10 conferencing system:

- Configure SIP Trunk server in IPVT10 server, which is the server address of Grandstream UCM.
- Configure VoIP Trunk, Outbound Route and other information in Grandstream UCM.

2. Calling out via UCM on IPVT10 server:

- Configure SIP Trunk server for calling out on IPVT10 server, which is the server address of Grandstream UCM.
- Configure Inbound Route in Grandstream UCM.

3. Introduce how to join into IPVideoTalk conferencing system for UCM users.



IPVIDEOTALK SERVICE CONFIGURATION ON UCM

Configure SIP Trunk on IPVT10

Login in IPVT10 Web UI, and access "SIP Trunk Service Configuration \rightarrow Access" to fill in the information below:

1. SIP Trunk Server Address (Necessary)

Accessible to any IP address: All IP addresses are allowed to access the IPVT10 server, including UCM SIP server.

Only accessible to the following IP address: Users need to fill in the UCM SIP server address, which means only the filled IP address is allowed to access the IPVT10 server.

 SIP Trunk Service Configuration - Access 		
SIP Trunk server address:	 Inaccessible 	
	 Accessible to any IP address 	
	• Only accessible to the following IP address	
	192.168.200.236	•
	Add more addresses	•

Figure 1: Configure SIP Trunk Server Address

2. IVR Service Access Number (Optional)

Grandstream UCM supports to dial into the IPVideoTalk meeting via meeting ID directly. Please kindly refer to the UCM User Guide for more details.

Configure Grandstream UCM

Configure VoIP Trunk

- 1. Login Grandstream UCM Web UI, and access to "Extension / Trunk \rightarrow VoIP Trunk".
- 2. Select Create New SIP Trunk, and fill the information into option "Provider Name" and "Host Name" as the figure shows below:





Menus		÷	Create New SIP Trunk	
a	System Status	~		
æ	Extension / Trunk	^	Type:	Peer SIP Trunk v
			* Provider Name:	IPVT10
			* Host Name :	192.168.126.242:5060
	Analog Trunks		Keep Original CID :	
	VoIP Trunks		Keep Trunk CID:	\Box
	SLA Station		NAT:	
	Outbound Routes		Disable This Trunk :	
			TEL LIRI:	Dicabled
Ś	Call Features	~		
⇔	PBX Settings	~	Caller ID :	
ç,	System Settings	~	CallerID Name :	
*	Maintenance	~	Auto Record :	
B	CDR	~	Direct Callback :	

Figure 2: Configure VoIP Trunk

- **Provider Name:** Users need to fill in the provider name, and the duplicated name is not allowed. The provider name will be shown up during inbound/outbound routing.
- Host Name: Users need to fill in the IP/URL address of IPVT10 server for Host Name option. If the IPVT10 server has customized port, users need to fill in the IP/URL address of IPVT10 server with the customized port, such as 192.168.126.242:5060.

Note:

Users need to make sure the SIP Transport and Port should be matched. For example, UDP/TCP protocol corresponds port number 5060, TLS protocol corresponds port number 5061.





 Service Port Configuration 			
Note: You can't set the custom ports as internal ports	View details		
Server components	Protocol	Port	Description
Meeting management server	HTTP	80	Meeting web page, management request, API server
Meeting management server	HTTPS/WSS	443	Meeting web page, management request, API server, websocket access
Signaling server	TCP/UDP	5060	Signaling access of devices, Trunk/PSTN docking
Signaling server	TLS	5061	Signaling access of devices, Trunk/PSTN docking
Media server	UDP	60000 ~ 65000	Media flow port range 🕢

Figure 3: Configure Service Port

VoIP Trunks				455% + 6.7K/s	
+ Create New SIP Trunk + Create New L	AX Trunk				
Provider Name 🗘	Terminal Type 🌻	Type 🌲	Hostname/IP 🌻	Username 🗘	Options
IPVT10	SIP	peer	192.168.126.242		r 🧐 💩 🛅
	6	iauro 4: VolD	Trunk Configuration		

Figure 4: VoIP Trunk Configuration

- 3. Configure VoIP Trunk SIP Transport and Codecs
 - SIP Transport: Users could select UDP/TCP/TLS as the SIP Transport, and the Port number should • correspond the SIP Transport type. Please see the figure below:

Menus 🗲	Edit SIP Trunk: IPVT10				
System Status Y	Basic Settings Advanced Setting	ngs			
🚑 Extension / Trunk 🔷					
Extensions	* Provider Name :	IPVT10]	* Host Name :	192.168.126.242:5060
Extension Groups	Auto Record :			Keep Original CID :	
Analog Trunks	Keep Trunk CID :			NAT:	
VoIP Trunks	Disable This Trunk:			TEL URI:	Disabled
SLA Station	Caller ID :			CallerID Name:	
Outbound Routes	From Domain :				
Inbound Routes	Transport:	UDP v		Direct Callback :	
🖉 Call Features 🗸 🗸			,		

Figure 5: Configure SIP Transport

- Codecs: Users need to select at least one codec which is supported by IPVideoTalk service. Otherwise, • the call cannot be established. The available codecs:
 - o Audio: "GSM / PCMU / PCMA / G.722 / OPUS"
 - Video: "H.264 / VP8" 0
- SRTP Mode: The default setting is "Disabled", and it is suggested to set as "Enabled but not forced". ٠





🗥 System Status	~	Basic Settings Advanced Sett	ings					
井 Extension / Trunk	^							
Extensions		Codec Preference :	12 items	Available			5 items	Selected
Extension Groups			Search	Q		Searc	h	Q
Analog Trunks			LBC	^	↑ ^		PCMU	^
VoIP Trunks			AAL2-G.726-32				GSM	
SLA Station			ADPCM	-	4		G.726	•
Outbound Routes		Send PPI Header:						
Inbound Routes		Send PAI Header :						
Call Features	~	Passthrough PAI Header:						
PBX Settings	~	DID Mode:	Request-line			~		
System Settings	~	DTMF Mode:	Default			~		
💥 Maintenance	~	Enable Heartbeat Detection :						
CDR	~	* The Maximum Number of Call Lines :	0					
Value-added Feature:	s v	Fax Mode:	None			~		
		SRTP:	Enabled but not forced			~]	
		Sync LDAP Enable :						

Figure 6: Configure SIP Transport and Codecs

Configure Outbound Route

Users could go to "Extension / Trunk → Outbound Routes", and click on "Add" to add the Outbound Route. As the figure shows below:

Menus 🗲	Edit Outbound Rule: xmeetings	_IPVT10		
System Status +				
🕂 Extension / Trunk 🔹	* Calling Rule Name :	xmeetings_IPVT10		
Extensions	* Pattern :	_*99x.		
Extensions				
Extension Groups				
Analog Trunks	Disable This Route :		PIN Groups:	None
VoIP Trunks	Password :		Privilege Level :	Internal
SLA Station	Enable Filter on Source Caller ID			Warning: Setting potential security
Outbound Routes	Enable Filter on Source Caller ID :			
Inbound Routes	Call Duration Limit			
🗳 Call Features 🗸 🗸	Call Duration Limit:			
🗘 PBX Settings 🗸 🗸	Main Trunk			
System Settings 🛛 🗸	* Trunk:	SIPTrunks IPVT10		
🗙 Maintenance 🗸 🗸 🗸	Strip:	3		
🖹 CDR 🗸 🗸	Prepend :			

- Figure 7: Configure Outbound Route
- Configure Calling Rule Name: Users need to fill in the Calling Rule Name for each Outbound Route, ٠ and the duplicated Calling Rule Name is not allowed.





Configure Pattern: Users need to configure "Pattern" to recognize the dialing numbers for UCM, and the initial pattern should be "_". The special characters and wildcard characters are allowed for patterns configuration. For instance, users could configure the pattern as "prefix + meeting ID", such as "_*99x". Then, UCM clients could dial "*99 + IPVideoTalk meeting ID" to dial into the meeting. The meeting ID could be 1 or multiple digits, and users may need to configure "Strip" option, please see the table below:

Parameters	Description
X/x	0-9
Z/z	1-9
N/n	2-9
[345-9]	3,4,5,6,7,8,9
!	0 or multiple characters (any character)
	1 or multiple characters (any character)

Table 1: Pattern Rule

- **Configure Privilege Level:** Users need to configure the VoIP Trunk Privilege Level as "Internal" since the UCM clients' default privilege level is "Internal". The privilege level of the clients should be no lower than Outbound Route privilege level. Otherwise, the server will send 603 error messages to the clients.
- **Configure Use Trunk:** Users need to select the configured VoIP Trunk.
- **Configure Strip:** Users could configure the how many characters will be ignored for the prefix. For example, if users want to "*99", users could set "3" for this option.

Users could click on "Save" \rightarrow "Apply" to create the new Outbound Route, as the figure shows below:

Outbound Routes				
An outgoing calling rule associates an low-cost SIP trunk. A failover trunk can	extension pattern with a trunk used to dial the patter be set up to be used when the primary trunk fails. N	m. This allows different patterns to be dialed through o tote: This panel only manages individual outgoing calli	different trunks. For example, "local" allows 7-digit diale ng rules.	d through FXO port while "long distance" allows 10-digit di
+ Add				
Sequence 🗘	Outbound Rule Name ≑	Pattern ≑	Privilege Level 🌲	Options
1	xmeetings_IPVT10	_*99x.	Internal	2 🖲 🗟 🛇 😒

Figure 8: Create Outbound Route

Configure UCM Clients

• **Configure Codecs:** Users could go to the UCM client's Web UI → Account → Codec Settings to select the codecs. Users have to select at least one same codec as the codec for the SIP account configured on the UCM clients.





Note: We recommend enabling the option "Use First Matching Vocoder in 200OK SDP".

G X V 3 2 7	5						🎓 Them	e 🕞 Reb	oot 📕 Exit
	Enterpri	se Phone	Admini	stration	Interfa	ice			English 🔻
	Status	Account	Advanced	Settings	Maintenand	e			
General Settings		Account 1	Account 2	Account 3	Account	4 Accoun	t 5 Acc	ount 6	()
Network Settings			DTM	15.	🗆 In audio	B DEC2022		50	Ĩ.
SIP Settings		DTM	DTN F Pavload Tvr	lF: be:	101	C RFC2033	U SIP IN	FU	- 11
Codec Settings		Pre	ferred Vocod	er:	Available		Se	lected	- H
Call Settings					G722 G729A/B G726-32 iLBC	× *		CMU A CMA pus	
		Preferre	d Video Code	ec :	Available H263	*	Se H	elected 264	
		Codec Neg	otiation Priori	ty :	Callee		٣	_	- 11
	Use First Ma	atching Vocoder	in 2000K SD)P :	🗹 Yes				- 11
		iL	BC Frame Siz	ze :	30 ms				- 11
		G726-3	32 ITU Payloa	ad :	2				
		G726-3	32 Dynamic F	PT :	126				- 11
		Opu	s Payload Typ	be :	123				- 11

Figure 9: Configure Codecs on Clients

• SRTP Mode: The default setting is "Disabled", and it is suggested to set as "Enabled but not forced".

H.263 Encoder Resolution :	CIF QCIF
SRTP Mode :	Enabled but not forced
SRTP Key Length :	AES 128&256 bit
Enable SRTP Key Lifetime :	✓ Yes
Silence Suppression :	I Yes
Voice Frames Per TX :	2

Figure 10: Configure SRTP Mode on Clients

Users also need to check the account configuration for the configuration on clients in order to make consistent for the account information.





Menus 🗲	Edit Extension: 2003							
C System Status 🗸	Basic Settings Me	dia Features	Specific Time	Follow Me				
🕂 Extension / Trunk 🔹 ^								
Extensions	SIP Settings							
Extension Groups	NAT:	~					* Can Direct Media:	Yes
Analog Trunks	* DTMF Mode:	RFC2833			~		* TEL URI:	Disabled
VoIP Trunks	* Alert-info:	None			~		* Fax Mode:	None
SLA Station	Fax to Email :	Yes			~		Enable T.38 UDPTL:	
Outbound Routes							SRTP:	 Enabled and forced
Inbound Routes	Strategy:	Allow All			~			
🗳 Call Features 🗸 👻	Codec Preference :	14	items	Available		3 items	Selected	
🛱 PBX Settings 🗸 🗸					> Sea	irch		
🗔 System Settings 🗸			L2-G.726-32	^		H.264		
🗶 Maintenance 🗸 🗸		AD	PCM 23			PCMU PCMA		
🖹 CDR 🗸 🗸		н.2	63	Į				
10 V 1 1 1 1								

Figure 11: Account Configuration on the UCM





CONFIGURE IPVIDEOTALK OUTBOUND CALLS VIA UCM

Configure Outbound Calls via SIP Trunk

Users could login IPVT10 Web UI, and go to "SIP Trunk Service Configuration \rightarrow Call", and fill the information for the options below:

1. Dial Prefix (Necessary)

This is used to recognize the numbers/characters for IPVT10 server, and IPVT10 server will forward the request to UCM server. Users have to fill in at least one special character (*#+) or word (a, b, c,), this character or word should be available on the dialing keypad.

Users also need to check option "Remove the prefix before dialing".

2. Target Server Address (Necessary)

Users have to fill in the UCM server address and SIP port for this option. Users need to make sure the SIP Transport and Port should be matched. For example, UDP/TCP protocol corresponds port number 5060, TLS protocol corresponds port number 5061.

3. SIP Transport (Necessary)

Users have to select one SIP Transport between TCP / UDP / TLS.

4. Authentication (Necessary)

Users need to select "No authentication needed for this platform".

SIP Trunk Service Configuration - Call		
Dial prefix:	*99	
	Remove the prefix before dialing	
	Tip: If dialing +86 123456, remove the prefix +86 and call 123456	5 to reach the third-party platform
Target server address:	192.168.200.30	5060
Protocol type:	TCP v	
Authentication:	• No authentication needed for this platform	
	O Unified platform accounts authentication	

Figure 12: Outbound Calls via SIP Trunk Configuration





Configure UCM Inbound routes

Users have to configure Inbound Route in Grandstream UCM. Users could go to "Extension / Trunk \rightarrow Inbound Routers", and select the created VoIP Trunk, click on "Add" to add Inbound Route, as the figure shows below:

Menus	•=	Inbound Routes				
System Status	Ý	+ Add	🗊 Blacklist	⊚ Set	Global Inbound Mode	
Extension / Trunk	^	Trunks:	SIP Trunks IPVT10			
Extensions						
Extension Groups			Pattern 🕈		CallerID Pattern ≑	
Analog Trunks						
SLA Station						
Outbound Routes						
Inbound Routes						

Figure 13: Add Inbound Route

Users need to fill in the "Inbound Route" information as the figure shows below:

	create New Inbound Nule			
🗥 System Status 🗸 🗸				
📇 Extension / Trunk 🔹 🔺			1	
Extensions	* Irunks:	SIPIrunks IPVITU V		
Extension Groups	* Pattern:	_*92x.	CallerID Pattern :	
Analog Trunks				
VolP Trunks	Disable This Route :		Prepend Trunk Name :	
SLA Station	Prepend User Defined Name :		Alert-info :	None
Outbound Routes	Allowed to seamless transfer:			
Inbound Routes	Dial Trunk:			
🗳 Call Features 🗸 🗸 🗸	Allowed DID Destination :	Extension ×		
🗘 PBX Settings 🗸 🗸	Inhound Multipla Moda:			
🗔 System Settings 🗸 🗸	moona maripie mode.			
🗶 Maintenance 🗸 🗸	Default Mode Mode 1			
E CDR v	* Default Destination :	By DID v		
📲 Value-added Features 🗸	Strip :	3		

Figure 14: Configure Inbound Route

• **Configure Trunk:** Users could select the created VoIP Trunk for this option.





- **Configure Pattern:** Users could configure "**_*99x.**" as the pattern, which means Inbound Route allows the numbers with prefix "*99". If the configuration is "**_x**", which means Inbound Route allows one or multiple digits calling number to dial into the meeting. The special characters and wildcard characters are allowed for this option and the rule is the same as Outbound Route configuration.
- **Configure Default Destination:** Users could configure default destination for the local clients. With this configuration, users could dial to UCM extensions directly via IPVideoTalk server.

Note:

When users dial to UCM via IPVideoTalk server, the SRTP Mode cannot be Disabled for VoIP Trunk, and it is suggested to set as "Enabled but not forced".





DIAL INTO IPVIDEOTALK MEETINGS

We assume client A has a registered IPVideoTalk ID, and client B has a registered SIP extension in UCM (e.g. 2003), the Dial Prefix for SIP Trunk is "*99".

UCM Extension Joins into IPVideoTalk Meeting

Scenario 1:

UCM extension joins into the IPVideoTalk meeting by dialing IPVideoTalk meeting ID via audio call.

Prerequisite:

Active meeting M

Operations:

Users could dial IPVideoTalk meeting ID M (*99M) to join into the meeting on client B.

IPVideoTalk Conferencing System Invites UCM Client to Join into the Meeting

• Scenario 1:

Schedule the IPVideoTalk meeting and invite client B, then start the IPVideoTalk meeting.

Operations:

- 1. Client A schedules and hosts the IPVideoTalk meeting M, set the SIP extension on client B (*992003) as the invitee.
- 2. The IPVideoTalk meeting M has been started.
- 3. Client B receives the incoming call and answers the call via audio mode.
- Scenario 2: The meeting host invites the client B during the IPVideoTalk meeting.
 Prerequisite: Client A is the meeting host during the IPVideoTalk meeting.
 Operations:
 - 1. Client A invites the SIP extension on client B (*992003) during the IPVideoTalk meeting.
 - 2. Client B receives the incoming call and answers the call via audio mode.

Conference Control

Prerequisite: Client A and client B are in the IPVideoTalk meeting M.

Operations:

- 1. Client A mutes/unmutes the audio.
- 2. Client B mutes/unmutes the audio.







Figure 15: Meeting Call using IPVideoTalk Service with the UCM

