# **CASE STUDY**

# MEET PANEL-SIP INTEGRATION

Description

This document describes how integrate a MEEN panel into a system based in SIP protocol, for receiving panel calls in one o more SIP telephones

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### INTRODUCTION

In some places (offices mainly), an existing SIP telephone is required for receiving calls from the MEET panel, instead of installing a new device in the receptionist's desk, such as a MEET monitor.

This Case Study explain how to configure a PABX based in a SIP server to receive calls from a MEET panel in a videotelephone.

This document is structured in two ways:

a) Panel's calls are received through the SIP server.

In this case, the calls can be received in several telephones at the same time. A SIP extension in the SIP server is required for each MEET panel.

b) Panel's calls are received directly in a videotelephone.
 In this case, only this videotelephone can receive and attend calls from MEET panel. No SIP server is required because the calls are going directly to the videotelephone.

The SIP server has to be able to manage the audio & video codecs that uses the panel: audio G.711 (PCMA-PCMU) and video H.264.

In this **Case Study** we have used a server GRANDSTREAM model UCM602 and a videotelephone SIP standard.

### MANAGING CALLS TROUGH A SIP SERVER

Panel's calls are managed from the SIP server, so it is mandatory to register the panel in the server as same as any SIP videotelephone

The steps are the following:

#### 1. Program the MEET panel to use SIP protocol

Go to "SIP" section in panel and program the following items:

- **SIP SERVER**: Indicate the IP of the SIP server in format: -> **sip:server\_ip**.
- **DOMAIN**: Indicate the IP server domain. -> sip\_server\_ip.
- **SIP USER:** Extension number that has to be used to register the panel in the SIP server.
- SIP PASS: Password related to SIP SERVER.
- SELECT THE BOX "ENABLE SIP".

	DOOR ENTI	RY SYSTEM	
DEVICE			
CENERAL			
NETWORK	ENABLE SIP:	SEARCH SIP STATUS	
ACCESS	SIP SERVER:	sip:192.168.1.190	
540141 05000	DOMAIN:	192.168.1.190	
FACIAL RECOG.	OUTBOUND:		
IP CAMERA	STUN IP:	192.168.1.233	
SIP	STUN PORT:	5060	
SIP TRUNK	H.264:	102	
	SIP USER:	111	
SIP CALL	SIP PASS:		
ADVANCED	CONVERSATION:	120s 🗸	
PINCODE	RING TIME:	30s 🗸	
QR ACCESS		SAVE	
RESET			
LOG OUT			

In this example we have assigned **USER SIP: 111**. This user number will be created as an extension number in the SIP server.

#### 2. Register the panel in the SIP server

• Go to the SIP section in the SIP webserver and select "Add".

5	UCM6202				Please go to the will be used to	e Change Binding Email retrieve your password	page to bind Email address to yo when you forget it.	ur account. This Email address		Q Арру С	unges English v 🛛 🕦 admin2v
		E	tensions								
		$\mathbf{C}$	A33 [	🖞 Edit 🗊 Delete 🛛 Reset	Ede All Sip Extensions	S Import S	Export 🗸 🔛 E-mail Not	fication @ Follow Me Opti	ons	Enter Extension	Number or CallerID Name Search
C	Extensions		<ul> <li>Status</li> </ul>	Presence Status	Extension 0	CallerID Name 0	Message	Terminal Type 0	IP and Port 0	Email Status 0	Options
	Extension Groups		e Ide	Available	171	telefono sip	Messages: 0/0/0	SIP(WebRTC)	192.168.1.162:5061	₽°o	🗹 🕫 🖞 🗖
							Total: 1 🔄 1 🖻				30 / page v Goto 1
в	CDR ~										

- Register the panel using the same data that has already used in the panel:
  - Extension -> The **SIP USER** assigned in the panel.
  - SIP/IAX Password -> The same used in **PASS SIP** of the panel.

Other information requested in this form could be required for the SIP server configuration, but it does not affect to the panel registration.

Finally, select "Save" and "Apply Changes""

⋝ UCM6202		Please go to the Change Binding Email page to b will be used to retrieve your password when you	oind Email address to your account. This Email address u forget it.	O Apply Changes English V   (1) adm
Menus 🖅	Create New Extension			Save Can
🗥 System Status 🗸 🗸	Basic Settings Media Fe	atures Specific Time Follow Me		
🕂 Extension / Trunk 🗠				
Extensions	* Select Extension Type :	SP Extension v		
Extension Groups	Select Add Method :	Single v		
Analog Trunks	General			
VoIP Trunks	• Extension :	111	CallerID Number :	
SLA Station	Permission:	Internal v	• SIPAAX Password :	123496abc
Outbound Routes	AuthID:		Voicemail:	Local Voicemail ~
Inbound Routes			Voicemeil Pessword:	1234
🕻 Call Features 🗸 🗸	Skip Voicemail Password Verification:		Send Voicemail to Email :	Default v
PBX Settings ~	Keep Voicemail after Emailing i	Default v	Enable Keep-alive I	
🖓 System Settings 🗸 🗸	Keep-alive Frequency:		Disable This Eccension :	
🗶 Maintenance 🗸 🗸			Fratte STA	
🖹 CDR 🗸 🗸	Servera Cille (D)		1	
Value-added Features v	consisting cars cars.			
	User Settings			
	First Name:		Last Name :	FERMAX PANEL
	Email Address :		User Password :	123456e
	• Language :	Default ~	Concurrent Registrations :	1
	Mobile Phone Number:			

Check that the panel (extension 111) has been correctly registered and appear as "Idle".

陭 UCM6202			Please go to th will be used to	e Change Binding Email pag retrieve your password whe	e to bind Email address to you in you forget it.	ir accourt. This Email address			Q English ~	() admin2~
Menus ·=	Extensions									
Fytension / Trunk	+ Add 🔯 Edit	10 Delete Reset	😂 Edit All Sip Extensions	S Import	port 🖌 🔀 E-mail Notil	Ication 🛛 🕹 Follow Me Optione				Search
Extensions	Status *	Presence Status #	Extension ¢	CallerID Name *	Message	Terminal Type 0	IP and Port #	Email Status +		Options
Extension Groups	· He	Available	111	FERMAX PANEL	Messages: 0/0/0	SIP(WebRTC)	192,168,1.111,5060	E	Ľ 9	Ċ 🗖
Analog Trunks	ldic • Idic	Available	171	telefono sip	Messages: 0/0/0	SIP(WebRTC)	192.168.1.162:5051	26	29	Ů 🗎
VoIP Trunks				-	otali 2 🔍 🧵 🗲				307 page V	Goto 1
SLA Station										
Outbound Routes										
Inbound Routes										
📞 Call Features 🔹										

It is possible that the indication "Idle" takes some time to appear.

#### NOTE:

Be sure to select the codecs G.711 (PCMA-PCMU) and H.264 in the SIP server.

Codec Preference :	14 items	Available	<	3 items	Selected
	Search	Q	>	Search	Q
	G.729	*	1	рсми	
	H.265		<u>^</u>	PCMA	
	iLBC		~	H.264	)
	RTX	•	4		

#### 3. Mapping the SIP call in panel

A mapping in panel is required to relate the number **dialled** in the keypad with the SIP extension number of the videotelephone that has to receive the call.

For example, if the videotelephone that has to receive the calls when someone calls to concierge (number 9901) in the extension number 171, a mapping from number 9901 to SIP extension number 171 is required.

In the case of a DIGITAL panel (MILO, KIN o MARINE MEET), it is necessary to use the SIP CALL function. Please follow instruction in the panel's INSTALLER MANUAL and make the corresponding mapping with the format: <a href="mailto:sip:ext\_number@server\_ip">sip:ext\_number@server\_ip</a>.

	DOOR E	NTRY SYSTER	M			
DEVICE					SIP CA	LL SETTINGS
GENERAL						
NETWORK	Seleccionar a	archivo Ninado IMPOR	EXPORT			
ACCESS	APARTMENT	NUMBER	APARTMENT	NUMBER	APARTMENT	NUMBER
FACIAL RECOG.	9901	sip:171@192.168.1.190				
IP CAMERA						
SIP						
SIP TRUNK						
SIP CALL						
ADVANCED						
PINCODE						

*In this example we have configured the MEET panel to generate a call to the extension 171 when someone calls the concierge (dialling number 9901 in panel).* 

In the case of MEET 1W panel, and based on the GENERAL factory settings parameters, that are as follow:

	D DOOR ENTRY S	BYSTEM
DEVICE		GENERAL SETTINGS
GENERAL		
NETWORK	TYPE:	1W PANEL 🗸
ACC	BLOCK:	1
CIP	APARTMENT:	101
SIF	DEVICE NO .:	1
SIP TRUNK	DEVICE TAG:	(≦16 CHARACTERS)
SIP CALL	LANGUAGE:	ENGLISH ¥
ADVANCED	PANEL VOLUME:	3
PINCODE	DOOR OPEN VOICE:	
	VIDEO RESOLUTION:	1280x720 V
RESTORE	SIP DIVERT MODE:	PARALLEL CALL
LOGOUT	TIME:	
	TIME ZONE:	GMT+01:00 V
		SAVE

NOTE:

The PANEL 1W pushbutton is related to APART 101 field in default mode.

The mapping has to be done to relate parameter **APART 101** with the **SIP extension number**, in format **sip: ext\_number@server\_ip.** 



*In this example we have programmed that when someone uses the pushbutton (associated to apartment 101) generates a call to the SIP extension number 171.* 

#### NOTE:

In the case of a call has to be received in several extensions, a group of extensions has to be programmed in the SIP server.

## CALLS RECEPTION WITHOUT SIP SERVER

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In the cases that a call has to be received in a unique videotelephone, the configuration can be easier, since no SIP server is required. Only make a mapping SIP call to point to the videotelephone **IP address** that has to receive the calls.

FERMA SISTEMA DE		RTERO ME	ET			
DISPOSITIVO	-			CO	NFIGURACIÓN LL	AMADA SIP
GENERAL						
CONFIG.RED	Seleccionar arch	nivo Ninado IMF	ORTAR EXPOR	TAR		
ACCESO	APARTAMENTO	NÚMERO	APARTAMENTO	NÚMERO	APARTAMENTO	NÚMERO
RECON. FACIAL	9901	sip:192.168.1.162				
CÁMARA IP						
SIP						
SIP TRUNK						
SIP CALL						
AVANZADO						
CÓDIGO PIN						

*In this example we have configured the MEET panel to generate a call to the telephone with IP 192.168.1.162 when someone call to the concierge (9901).* 

FERMA SISTEMA DE		RTERO MEE	т		
DISPOSITIVO				CONFIGUR	ACION LLAMAD
GENERAL					
CONFIG.RED	APART.:	101			
ACC	NÚMERO:	sip:192.168.1.162			
SIP	BORRAR:	GUARDAR			
SIP TRUNK					
SIP CALL	APARTAMENT	O NÚMERO	APARTAMENTO	NÚMERO AF	PARTAMENTONÚN
AVANZADO	101	sip:192.168.1.162			
CÓDIGO PIN					
RESTAURAR					
CERRAR SESIÓN					

*In this example we have configured the MEET 1W panel to generate a call to the telephone with IP 192.168.1.162 when someone press the pushbutton.*