



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:

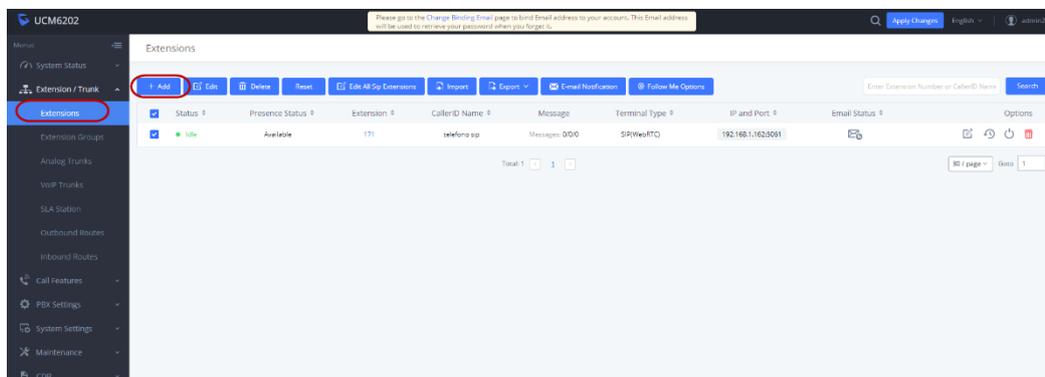
- **SIP SERVER:** Indicate the IP of the SIP server in format: -> **sip:server_ip**.
- **DOMAIN:** Indicate the IP server domain. -> **sip_server_ip**.
- **SIP USER:** Extension number that has to be used to register the panel in the SIP server.
- **SIP PASS:** Password related to SIP SERVER.
- SELECT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



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

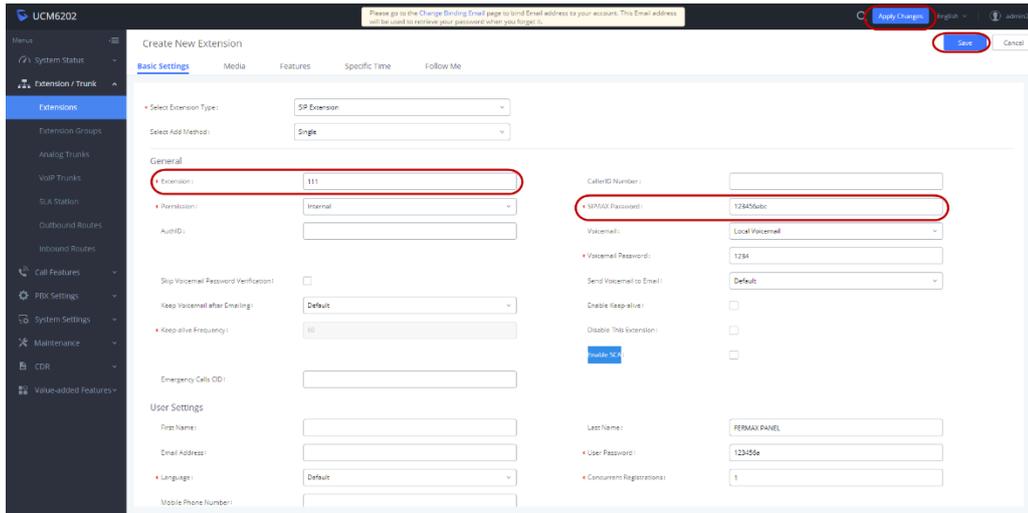
- Go to the SIP section in the SIP webserver and select “Add”.



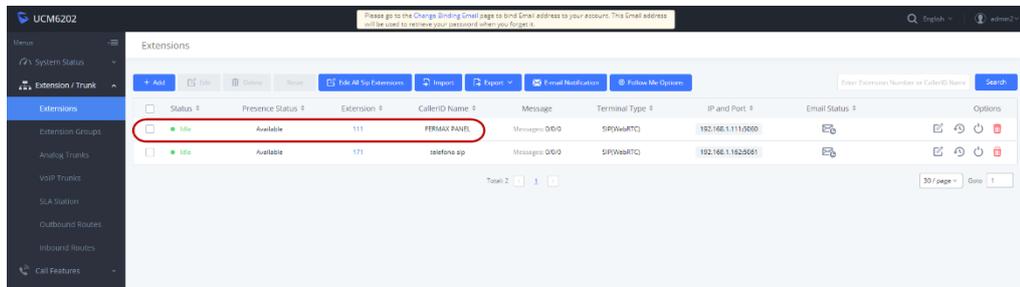
- Register the panel using the same data that has already used in the panel:
 - Extension -> The **SIP USER** assigned in the panel.
 - SIP/IAX Password -> The same used in **PASS SIP** of the panel.

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”



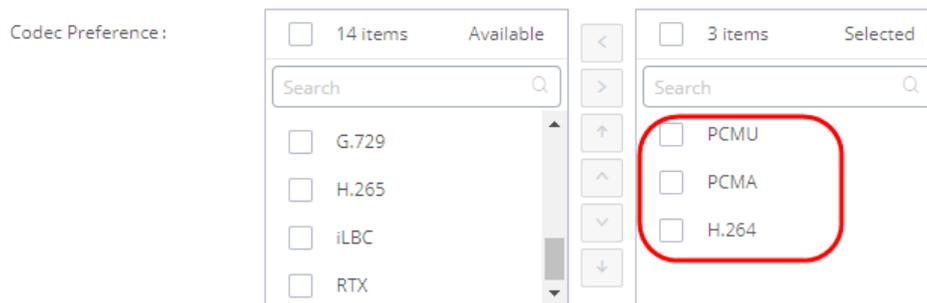
Check that the panel (extension 111) has been correctly registered and appear as “Idle”.



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.



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: [sip:ext_number@server_ip](#).

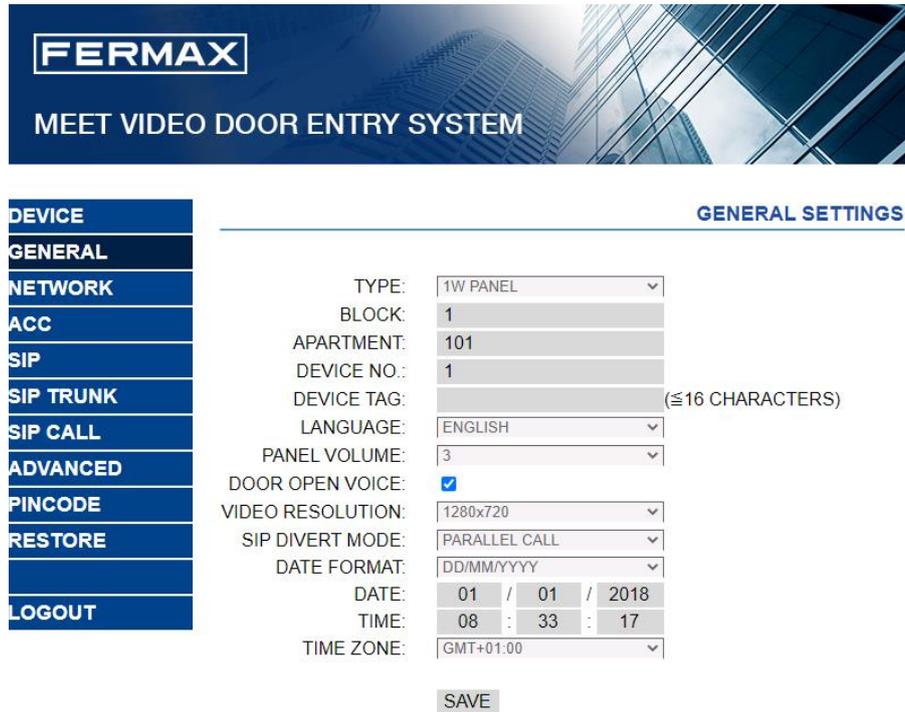


The screenshot displays the 'SIP CALL SETTINGS' configuration page for the FERMAX MEET VIDEO DOOR ENTRY SYSTEM. On the left is a vertical menu with the following items: DEVICE, GENERAL, NETWORK, ACCESS, FACIAL RECOG., IP CAMERA, SIP, SIP TRUNK, SIP CALL (highlighted), ADVANCED, and PINCODE. The main content area features a header with 'SIP CALL SETTINGS' and a toolbar with 'Seleccionar archivo', 'Nin...ado', 'IMPORT', and 'EXPORT' buttons. Below the toolbar is a table with the following data:

APARTMENT	NUMBER	APARTMENT	NUMBER	APARTMENT	NUMBER
9901	sip:171@192.168.1.190				

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:



FERMAX
MEET VIDEO DOOR ENTRY SYSTEM

GENERAL SETTINGS

TYPE:	1W PANEL	▼
BLOCK:	1	
APARTMENT:	101	
DEVICE NO.:	1	
DEVICE TAG:		(≤16 CHARACTERS)
LANGUAGE:	ENGLISH	▼
PANEL VOLUME:	3	▼
DOOR OPEN VOICE:	<input checked="" type="checkbox"/>	
VIDEO RESOLUTION:	1280x720	▼
SIP DIVERT MODE:	PARALLEL CALL	▼
DATE FORMAT:	DD/MM/YYYY	▼
DATE:	01 / 01 / 2018	
TIME:	08 : 33 : 17	
TIME ZONE:	GMT+01:00	▼

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**.

FERMAX
SISTEMA DE VIDEOPORTERO MEET

CONFIGURACION LLA

DISPOSITIVO
GENERAL
CONFIG.RED
ACC
SIP
SIP TRUNK
SIP CALL
AVANZADO
CÓDIGO PIN
RESTAURAR
CERRAR SESIÓN

APART.: 101
NÚMERO: sip:171@192.168.1
BORRAR:
GUARDAR

APARTAMENTO	NÚMERO	APARTAMENTO	NÚMERO	APARTAMENTO
101	sip:171@192.168.1.190			

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

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.

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).

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.