

# CASE STUDY

## Integration of Meet Intercom system with DorSIP PBX from DASSnet DORLET

### Description

It describes how to configure the Meet outdoor panel and the DorSIP PBX from DASSnet to route calls from the panel to any SIP extension.

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## INTRODUCTION

In offices and companies, it is common to connect the video door entry panel to the telephone switchboard to receive the call on a specific extension, preferably on the employee's computer, simplifying the work of installing and receiving calls. With the evolution of switchboards, a world of previously unimaginable advantages is now available, such as receiving the call on any extension located in different locations or even on the smartphone, and receiving video on the call, which was not available on an analogue telephone interface. You therefore have the same features as in a video door entry monitor if the terminal receiving the call has a screen: two-way conversation, visualisation of the visit and door opening.

The Fermax outdoor panel has evolved to adapt to this environment, and a clear example is the MEET panel, with IP technology, which supports SIP protocol, necessary in this type of integration. In this application, the outdoor panel must register in the SIP switchboard with a username and password and route calls to the required extension(s).

In the specific case of the DORLET DASSnet® SIP client, let's see how to configure the MEET outdoor panel, the DorSIP PBX and the DASSnet® client so that they can work together.

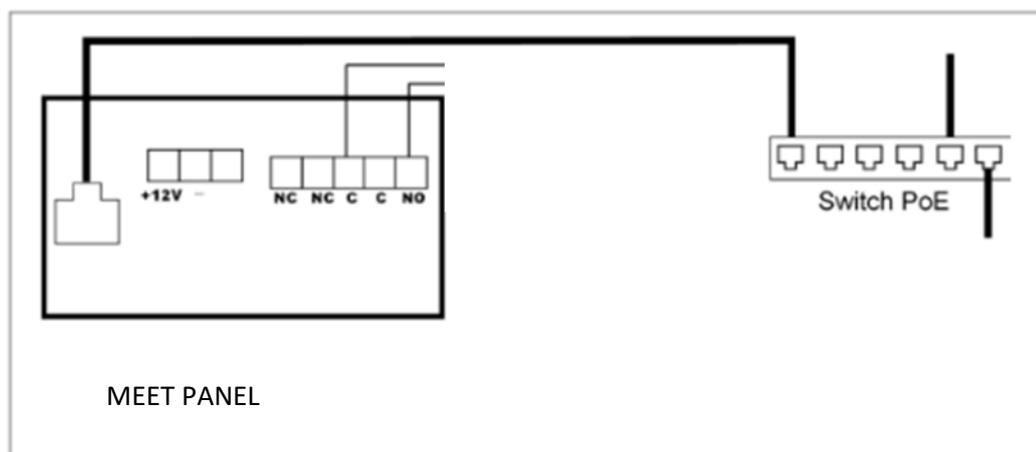
## MATERIAL REQUIRED

Any Fermax MEET outdoor panel can be used for this integration, as all of them support SIP protocol. If you want to call a single extension, it is advisable to use the one-way MILO panel (a single pushbutton) or the KIN or MARINE panel in one-way mode. If you want to call different extensions to locate different people, you can use the MILO panel or, if an electronic directory is required, the KIN or MARINE panel.

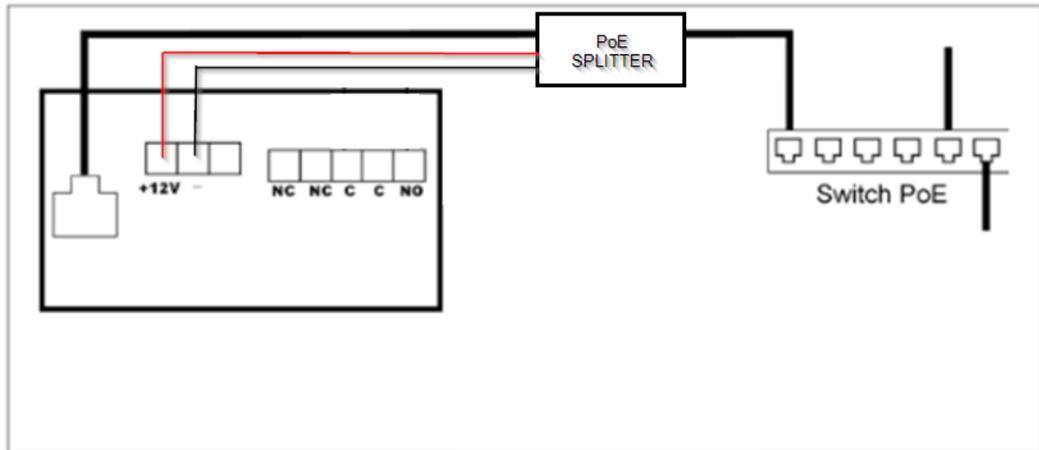
The outdoor panel may require a power supply if PoE is not supported.

A DORLET controller (UCA) must be used for door unlock as the door opener will be connected to it. The Mifare proximity reader of the outdoor panel (KIN, MARINE, MILO building) will be connected to the controller for full integration.

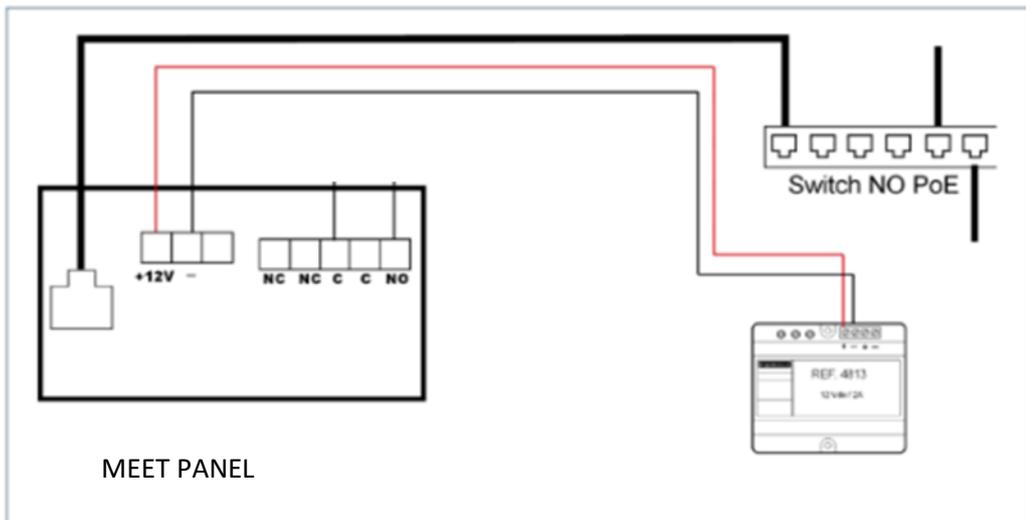
## BASIC SCHEME



Power supply to the panel via PoE switch.

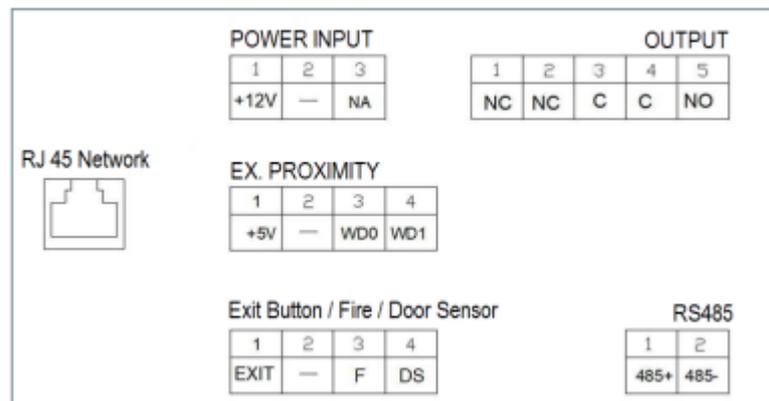


Power supply to the panel via PoE splitter.



Power supply to the panel via additional power supply.

Connection of the outdoor panel to the controller (UCA):



The EX.PROXIMITY connector contains the Wiegand-26 signals from the internal Mifare proximity reader:

- WD0= Data0
- WD1= Data1
- - = negative.

If the door-open voice message is to be played on the outdoor panel, the UCA shall make a contact closure between the EXIT and - signals when allowing access to users.

## INSTALLATION

Once the outdoor panel is installed, it must be connected to a switch or directly to the installation's router. The door opener connection will be wired to the UCA. Finally, the panel will be wired to the power supply or PoE splitter if required.

It is possible to connect a MEET monitor to simultaneously receive the call inside the office, as a backup to the call through the switchboard.

## CONFIGURATION

### SIP PABX

The first thing we will have to do for the correct operation of the MEET outdoor panel in DASSnet® is to configure the different extensions necessary for its correct operation in the SIP PBX. Any SIP PBX can be used.

In the case of the DorSIP switchboard, registration is carried out as follows:

- 1) Create an extension for the outdoor panel (6011 in the example), under [PBX Configuration - Extensions](#).

- Básico
- Extensiones
- Control de Llamadas entrantes
- Grupos de Timbrado
- Opción
- IssabelPBX sin Embeber

## Extensión: 6011

[Eliminar extensión 6011](#)  
 Usado como destino por 1 objeto:  
[Add Follow Me Settings](#)  
[Add Gabcast Settings](#)  
[- Editar extensión](#)

---

Nombre para mostrar

CID Num

Alias

Alias SIP

- Opciones de la extensión

---

CID saliente

Asterisk Dial Options   Override

Ring Time

Call Forward Ring Time

Outbound Concurrency Limit

Llamada en espera

Internal Auto Answer

Call Screening

Pinless Dialing

CID de emergencia

- Assigned DID/CID

2) We create an extension for the DASSnet server (6002 in the example). This extension is necessary in case you want to perform automatic actions.

- Básico
- Extensiones
- Control de Llamadas entrantes
- Grupos de Timbrado
- Opción
- IssabelPBX sin Embeber

## Extensión: 6002

[Eliminar extensión 6002](#)  
[Add Follow Me Settings](#)  
[Add Gabcast Settings](#)  
[- Editar extensión](#)

---

Nombre para mostrar

CID Num Alias

Alias SIP

- Opciones de la extensión

---

CID saliente

Asterisk Dial Options   Override

Ring Time

Call Forward Ring Time

Outbound Concurrency Limit

Llamada en espera

Internal Auto Answer

Call Screening

Pinless Dialing

CID de emergencia

- Assigned DID/CID

---

Descripción del DID

Añadir DID entrante

Añadir CID saliente

- Opciones del dispositivo

---

Este dispositivo usa la tecnología sip.

secret

dtmfmode

- 3) We create another extension for the receiver of the call, the DASSnet client (6008 in the example).

The screenshot shows the Asterisk web interface for configuring extension 6008. On the left is a navigation menu with options: Básico, Extensiones, Control de Llamadas entrantes, Grupos de Timbrado, Opción, and IssabelPBX sin Embeber. The main header reads 'Extensión: 6008' and includes links for 'Eliminar extensión 6008', 'Add Follow Me Settings', 'Add Gabcast Settings', and '- Editar extensión'. The configuration is divided into three sections: 'Opciones de la extensión', 'Opciones del dispositivo', and 'Opciones del dispositivo'. The first section contains fields for 'Nombre para mostrar' (Cliente DASSnet pruebas), 'CID Num Alias', and 'Alias SIP'. The second section contains various call handling options: 'CID saliente' (Cliente DASSnet), 'Asterisk Dial Options' (tr), 'Ring Time' (Por defecto), 'Call Forward Ring Time' (Por defecto), 'Outbound Concurrency Limit' (No Limit), 'Llamada en espera' (Habilitar), 'Internal Auto Answer' (Deshabilitar), 'Call Screening' (Deshabilitar), and 'Pinless Dialing' (Deshabilitar). The third section contains 'CID de emergencia' and 'Assigned DID/CID'. The bottom section, 'Este dispositivo usa la tecnología sip.', includes 'secret' (secret) and 'dtmfmode' (RFC 2833).

- 4) Finally, we create a call group (60000 in the below example), where we configure the two extensions the door panel shall call simultaneously.

## Grupo de extensiones: 60000

[Eliminar grupo de extensiones](#)

Editar grupo de extensiones

---

Descripción del grupo de extensiones: Prueba AXIS

Ring Strategy: Sonar todos

Ring Time (max 300 sec): 20

Lista de extensiones: 6008  
6002

Selector rápido de extensiones: (Seleccione una extensión)

Anuncio: Ninguno

¿Reproducir música en espera?: Sonar

CID Name Prefix:

Información de alerta:

Ignore CF Settings:

Ignorar agentes ocupados:

Enable Call Pickup:

Confirmar llamadas:

Anuncio remoto: Por defecto

Too-Late Announce: Por defecto

---

Change External CID Configuration

Mode: Por defecto

Fixed CID Value:

Call Recording

---

Record Calls: [Always](#) [On Demand](#) [Never](#)

Destino si no hay respuesta:

Terminar llamada:  Colgar:

### OUTDOOR PANEL CONFIGURATION

To configure the outdoor panel, use a browser and access the IP address assigned to the entrance panel, by default 10.1.0.1 (building entrance panel) or 10.1.1.2 (one-way panel) and enter the user admin and default password 123456.

The following steps will be followed:

#### 1) General configuration

##### a. One-way panel

You must configure the panel as an Individual Panel, assign a block (default 1), a dwelling (default is 101), and a panel number (default 1).

The Device Tag is important to identify the origin of the call on SIP extensions. The resolution of the camera shall be adjusted to the needs of the terminals used to receive the call.



DEVICE	GENERAL SETTINGS	
<b>GENERAL</b>	TYPE:	1W PANEL
NETWORK	BLOCK:	1
ACC	APARTMENT:	101
SIP	DEVICE NO.:	1
SIP TRUNK	DEVICE TAG:	FERMAX (≤16 CHARACTERS)
SIP CALL	LANGUAGE:	ENGLISH
ADVANCED	PANEL VOLUME:	5
PINCODE	DOOR OPEN VOICE:	<input checked="" type="checkbox"/>
RESTORE	VIDEO RESOLUTION:	640x480
	SIP DIVERT MODE:	PARALLEL CALL
	DATE FORMAT:	DD/MM/YYYY
	DATE:	01 / 01 / 2018
	TIME:	12 : 34 : 07
	TIME ZONE:	GMT+01:00
	<b>SAVE</b>	

b. Building Panel

In this example we have used a KIN building entrance panel. The panel must be configured as a Block Panel, assigned a block (default 1) and a panel number (default 1). The Device Tag is important to identify the origin of the call on SIP extensions. The resolution of the camera shall be adjusted to the needs of the terminals used to receive the call.

DEVICE	GENERAL SETTINGS	
<b>GENERAL</b>	TYPE:	BLOCK PANEL-DIGITAL
NETWORK	BLOCK:	1
ACCESS	DEVICE NO.:	1
FACIAL RECOG.	DEVICE TAG:	FERMAX (≤16 CHARACTERS)
LIFT	ALPHANUMERIC	<input type="checkbox"/>
IP CAMERA	KEYPAD:	<input type="checkbox"/>
SIP	LANGUAGE:	ENGLISH
SIP TRUNK	PANEL VOLUME:	3
SIP CALL	BACKGROUND:	DARK COLOR
ADVANCED	BRIGHTNESS:	250
PINCODE	VOICE SYNTH.:	<input checked="" type="checkbox"/>
WECHAT QR	VIDEO	
RESET	RESOLUTION:	640x480
	SIP DIVERT MODE:	PARALLEL CALL
	SCREENSAVER:	<input type="checkbox"/> (PNG,600*1024)
		<input type="button" value="Seleccionar archivo"/> <input type="button" value="Nin...lec."/> <input type="button" value="IMPORT"/> <input type="button" value="EXPORT"/> <input type="button" value="DELETE"/>
	HELP:	<input checked="" type="checkbox"/> (PNG,600*1024)
		<input type="button" value="Seleccionar archivo"/> <input type="button" value="Nin...lec."/> <input type="button" value="IMPORT"/> <input type="button" value="EXPORT"/> <input type="button" value="DELETE"/>

In case of calling more than one extension at the same time, it must be indicated whether the call will be made in parallel or sequentially, calling the next extension if the previous one is not answered within 30 seconds.

2) Network settings

An IP address compatible with the installation's lan will be assigned and the Gateway or router address will be indicated in order to have access to the Internet if you wish to make the call forwarding to the mobile phone. It is necessary to define a DNS server.

The IP address of the software will be left as the default one, unless it is required to manage access control registrations and cancellations for employees (proximity cards or facial recognition). In this case, the IP address of the computer on which the MEET management software (MMS) is installed shall be indicated.



The screenshot shows the FERMAX MEET VIDEO DOOR ENTRY SYSTEM interface. On the left is a vertical menu with options: DEVICE, GENERAL, NETWORK (highlighted), ACCESS, FACIAL RECOG., LIFT, IP CAMERA, SIP, SIP TRUNK, SIP CALL, ADVANCED, PINCODE, WECHAT QR, and RESET. The main area is titled 'NETWORK SETTINGS' and contains the following configuration fields:

IP:	192.168.1.224
MASK:	255.255.255.0
GATEWAY:	192.168.1.1
DNS:	8.8.8.8
SOFTWARE IP:	192.168.1.223
SW. PIN:	.....

Below the fields is a 'SAVE' button.

### 3) SIP PBX configuration

In the SIP Configuration section, the IP of the SIP PBX will be indicated, as shown in the screen below (192.168.1.221). Enter the user (extension number) and the password assigned to the outdoor panel extension (registration password). Next, check if the panel is correctly configured in the switchboard by clicking on the VIEW SIP STATUS link. If REGISTERED does not appear, check the information entered.



DEVICE	SIP SETTINGS
GENERAL	
NETWORK	ENABLE SIP: <input checked="" type="checkbox"/> <a href="#">SEARCH SIP STATUS</a>
ACCESS	SIP SERVER: <input type="text" value="sip:192.168.1.191"/>
FACIAL RECOG.	DOMAIN: <input type="text" value="192.168.1.191"/>
LIFT	OUTBOUND: <input type="text"/>
IP CAMERA	STUN IP: <input type="text"/>
SIP	STUN PORT: <input type="text" value="5060"/>
SIP TRUNK	H.264: <input type="text" value="102"/>
SIP CALL	SIP USER: <input type="text" value="6011"/>
ADVANCED	SIP PASS: <input type="text" value="....."/>
PINCODE	CONVERSATION: <input type="text" value="120s"/>
WECHAT QR	RING TIME: <input type="text" value="30s"/>
RESET	<input type="button" value="SAVE"/>

- **SIP SERVER:** sip: IP of the PBX
- **DOMAIN:** IP address of the PBX
- **STUN PORT:** UPD port of the control unit, 5060
- **SIP USER:** Extension that we have given to the intercom within the switchboard.
- **PASS SIP:** Password that we have configured in the PBX to the extension.

4) Call extensions configuration

a. One-way Panel

Go to the 'SIP CALL' tab and enter the flat number indicated in the 'GENERAL' section (101 in our example) and as NUMBER the SIP extension you want to call (6008 in the example) and click on 'SAVE'.

DEVICE	SIP CALL SETTING
GENERAL	
NETWORK	APARTMENT: <input type="text" value="101"/>
ACC	NUMBER: <input type="text" value="sip:6008@192.168.1.191"/>
SIP	DELETE: <input type="checkbox"/>
SIP TRUNK	<input type="button" value="SAVE"/>
SIP CALL	
ADVANCED	
PINCODE	
RESTORE	

APARTMENT	NUMBER	APARTMENT	NUMBER	APARTMENT	NUMBER
101	sip:6008@192.168.1.191				

b. Building panel

A CSV file will be configured with the assignment of call codes to the extensions to be called. The CSV file format can be downloaded from the SIP CALL section of the panel's web server, EXPORT option.

An example is the following:

APARTMENT,NUMBER,			
1,sip:6008@192.168.1.191			
5,sip:6009@192.168.1.191			
3,sip:6010@192.168.1.191,			
4,sip:6011@192.168.1.191;sip:00879892@sip.fermax.com,			

First indicate the code to be dialled and separated by a comma, the extension of the switchboard that is to receive the call when this code is dialled. In the example, dialling 1 will call extension 6008. Dialling 2 will call extension 6009.

It is possible to call more than one extension by listing several extensions in the same row separated by semicolons. One of these extensions can be a MeetMe licence to be able to call a smartphone via the MeetMe application (example 4). This allows calling a smartphone outside the local network, including video.

To load the CSV file on the panel, select the file created and click on the IMPORT button. The list of calls of the file will appear on the screen. Sometimes it is necessary to refresh the browser screen by clicking on the corresponding icon:



DEVICE

GENERAL

NETWORK

ACCESS

FACIAL RECOG.

LIFT

IP CAMERA

SIP

SIP TRUNK

SIP CALL

ADVANCED

Seleccionar archivo Nin...lec. IMPORT EXPORT

APARTMENT	NUMBER	APARTMENT	NUMBER
1	sip:1003@192.168.1.191;	5	sip:1008@192.168.1.
4	sip:192.168.1.243;sip:00879892@sip.fermax.com		

If we want to call by means of the agenda in the case of a KIN panel, the agenda will be created in a CSV file (different from the previous one). The CSV file format can be downloaded from the ADVANCED section of the panel's web server, EXPORT option.

One row is filled in per phonebook item, indicating the call code, Name to be displayed, blank, Y,. Example:

APARTMENT,NAME,MAPPING CODE,WHITELIST(Y),			
19,CARLOS FERRER, ,Y,			
12,PEPE GARCIA, ,Y,			
13,ANDRES LOPEZ,,Y,			
14,Alicia MARTINEZ,,Y,			

The file is loaded using the IMPORT option:

The screenshot shows the FERMAX MEET VIDEO DOOR ENTRY SYSTEM interface. On the left is a vertical menu with options: DEVICE, GENERAL, NETWORK, ACCESS, FACIAL RECOG., LIFT, IP CAMERA, SIP, SIP TRUNK, SIP CALL, ADVANCED (highlighted), PINCODE, WECHAT QR, and RESET. On the right, under 'ADVANCED SETTINGS', there are several configuration options: QUICK DIAL (checkbox), URL (text field), ONU(GPON) (checkbox), MAPPING CALL (checkbox), WHITE LIST (checkbox), and DIRECTORY (checkbox, which is checked). Below these options are buttons for 'Seleccionar archivo', 'Nin...lec.', 'IMPORT', and 'EXPORT'. A 'SAVE' button is located at the bottom of the settings area.

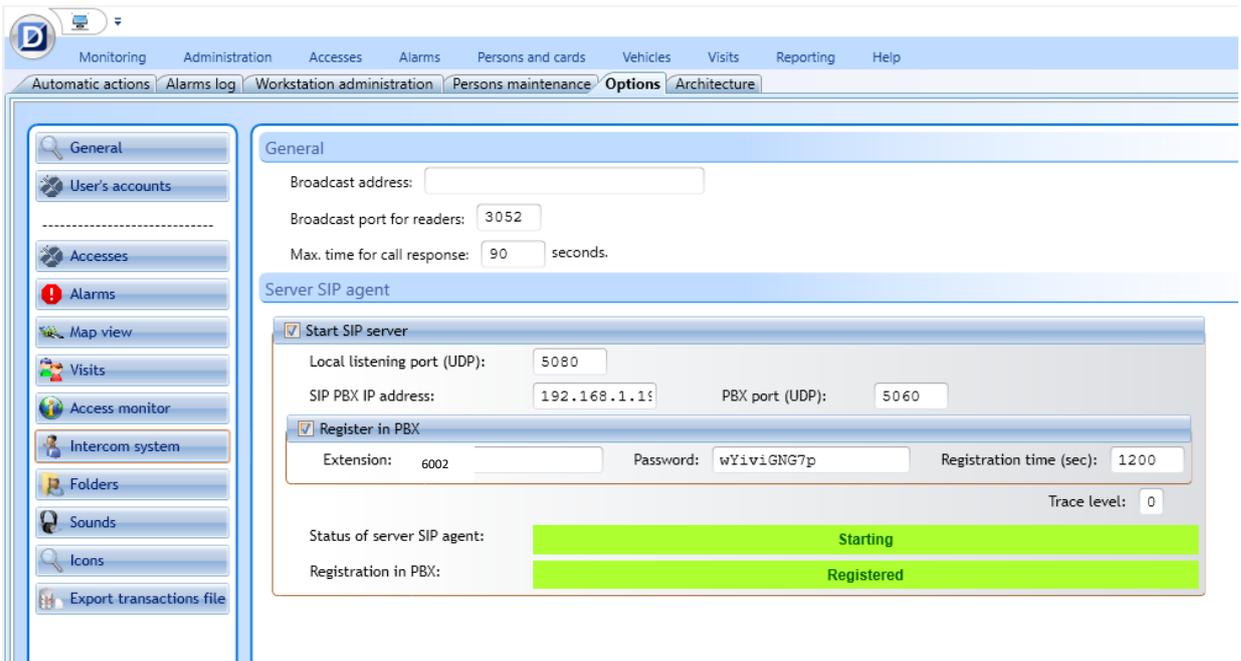
The DIRECTORY option is enabled so that this option appears on the panel.

### DASSnet® Client CONFIGURATION

Within the DASSnet® client we have to configure several sections.

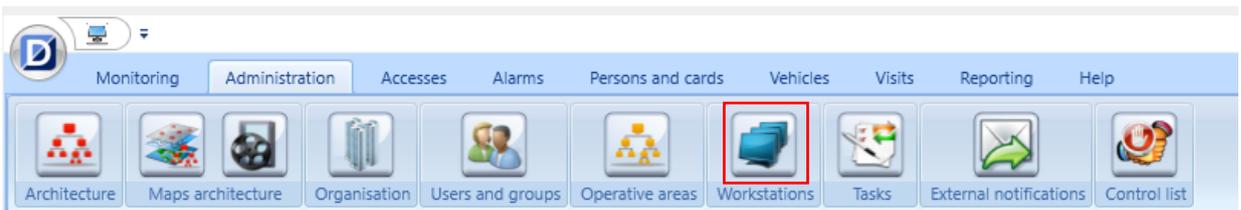
#### 1) SIP Agent Server

Go to options and in the **intercom** section, configure the SIP Agent server.

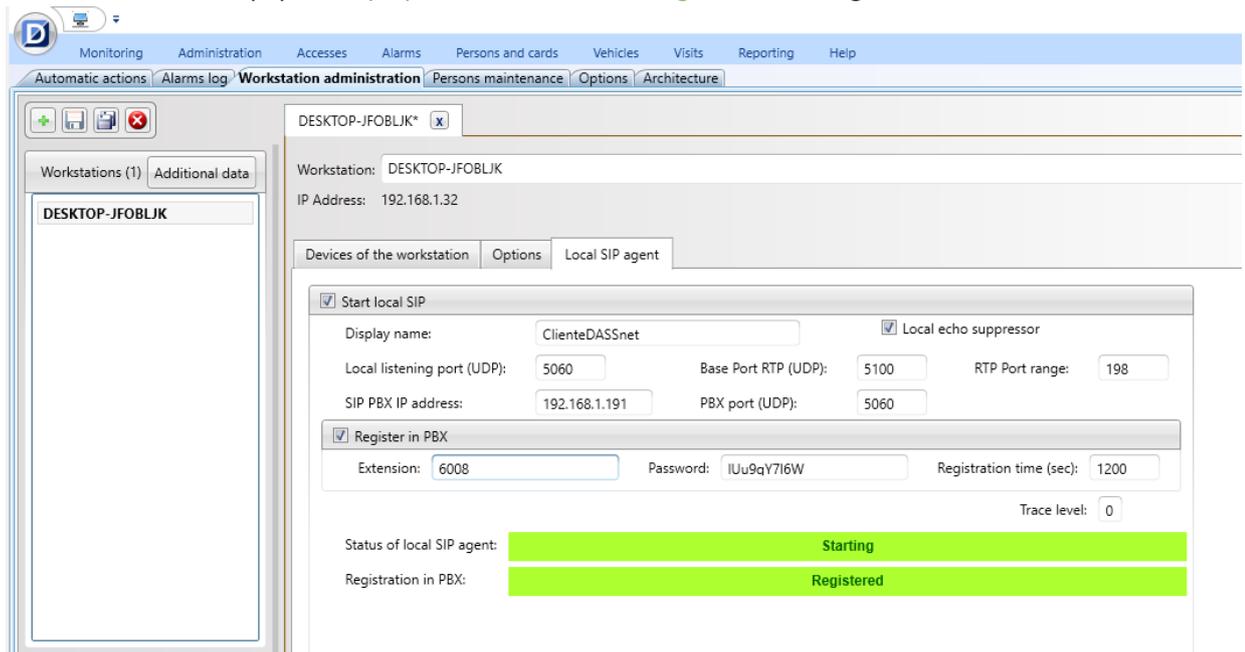


## 2) Local SIP Agent

Go to Administration - Workstations

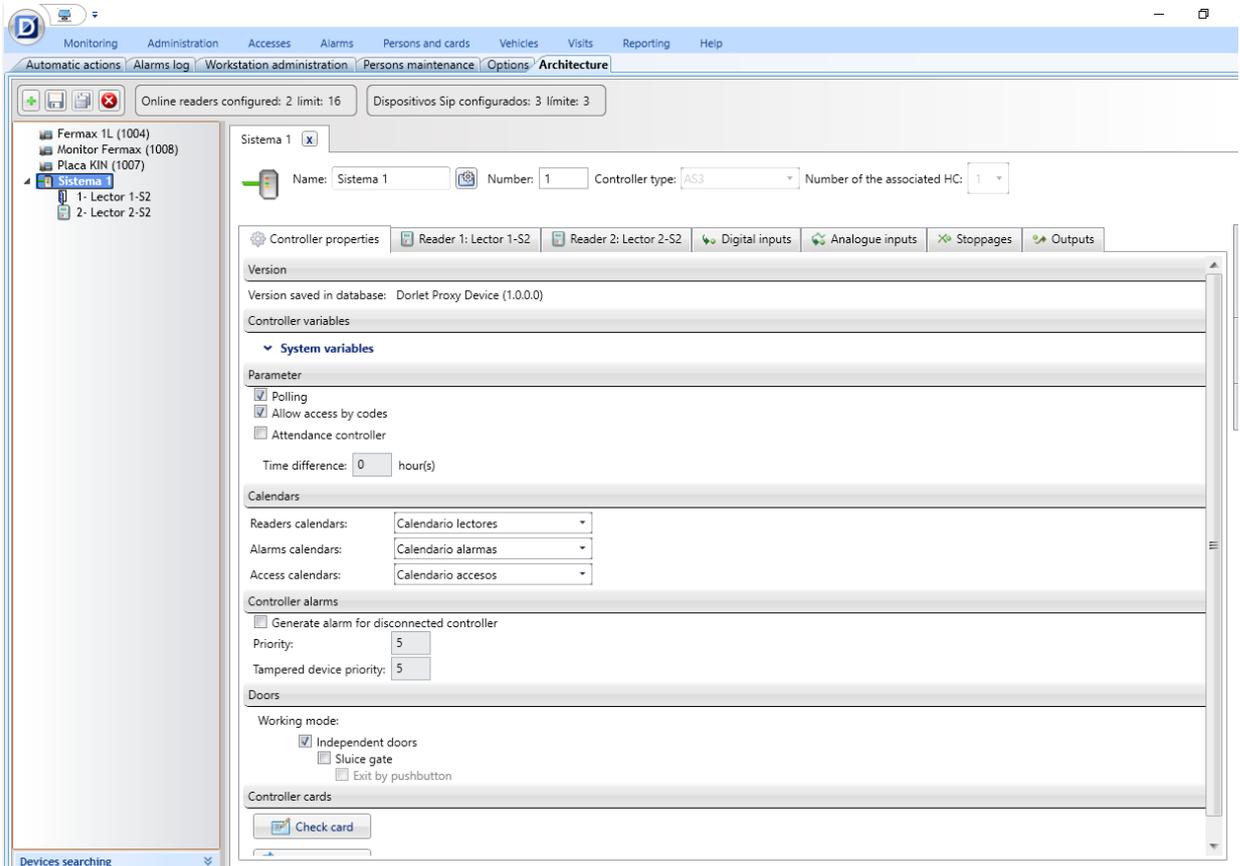


Select the equipment (PC) and in the **Local SIP Agent** tab, configure the PBX data.

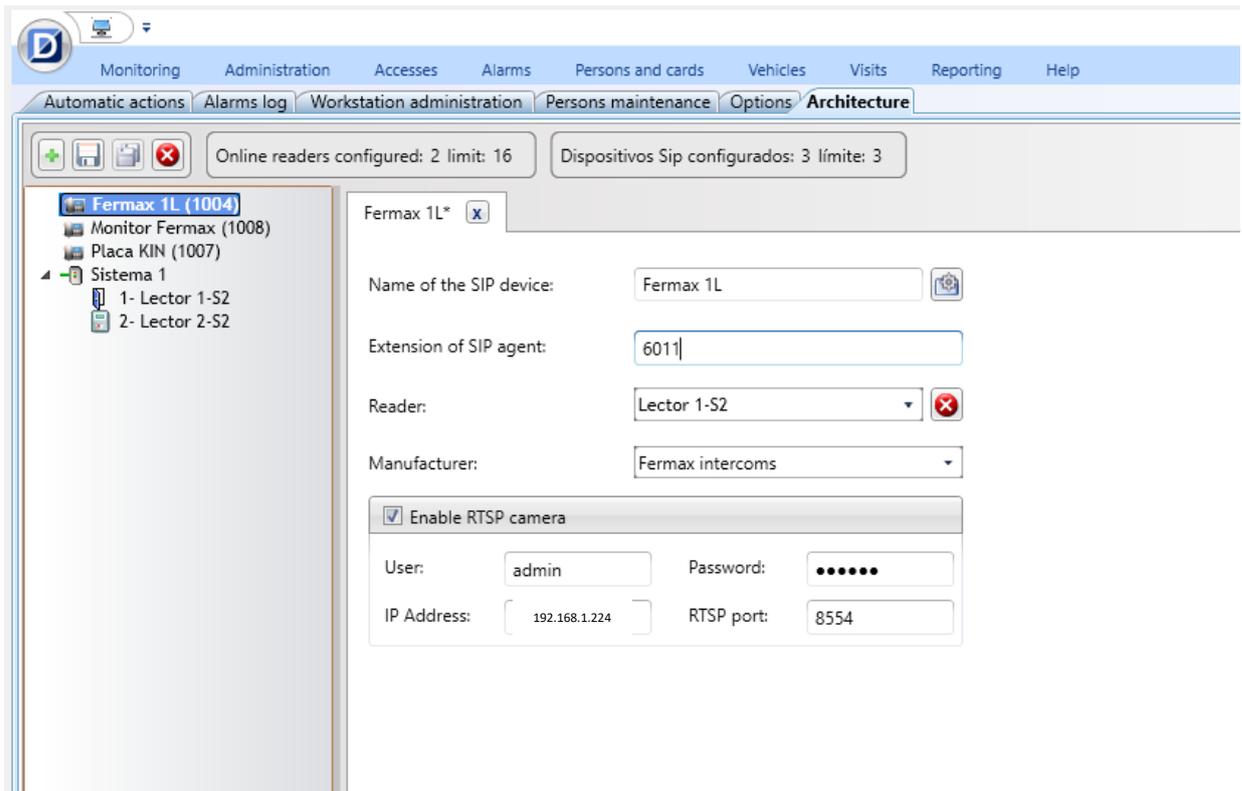


### 3) Architecture

We create a **system** and then assign the reader to the outdoor panel in order to be able to control some actions from the outdoor panel.

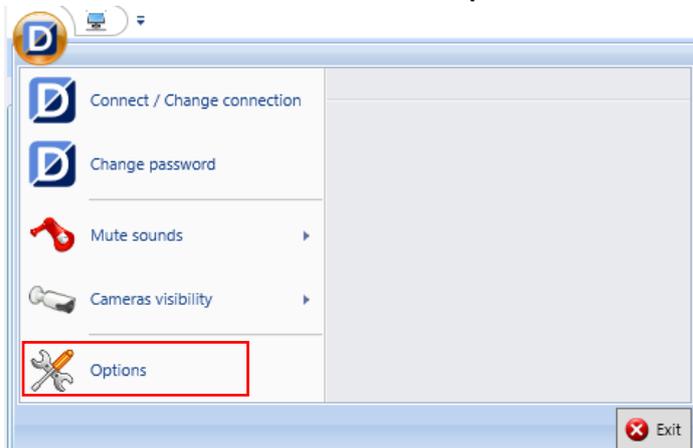


Create a new **SIP device** (+ New SIP device)

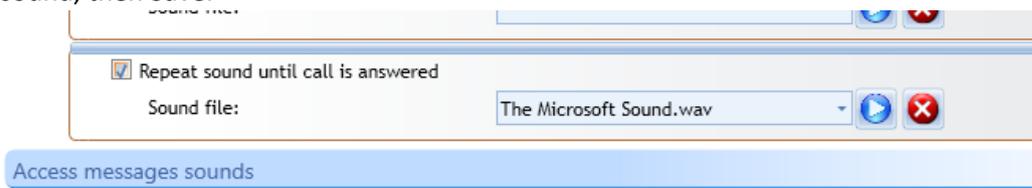


- **SIP device name** : The name of the intercom within DASSnet®.
- **SIP device extension** : The extension previously assigned to the outdoor panel in the PBX.
- **Reader** : This section is optional, but if we assign a reader we will be able to carry out several actions on the reader from the intercom.
- **Manufacturer** : In our case we will select Fermax
- Select **Enable RTSP camera** and fill in the details
  - **User** : User for accessing the outdoor panel settings
  - **Password** : Password for accessing the outdoor panel settings
  - **IP address** : ip address of the panel
  - **RTSP port** : Port for the video, in this case 8554.

A call sound assigned to the outdoor panel must be configured so that a melody is played when the call is received. This is done from the **options** section:



Sounds section. In 'repeat sound', activate the option "until answer call" and choose the desired sound, then Save.

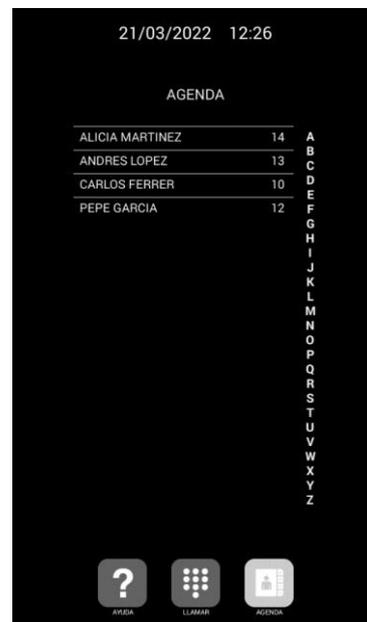


## OPERATION

To call a specific extension, press the call button (one-way panel) or enter the associated call code (building panel) and confirm with the bell button (Milo, Marine) or the off-hook icon (KIN). In the case of the KIN or MARINE panel, you also have the option of making the call via the Phonebook, by searching for the name of the receiver and pressing on it.

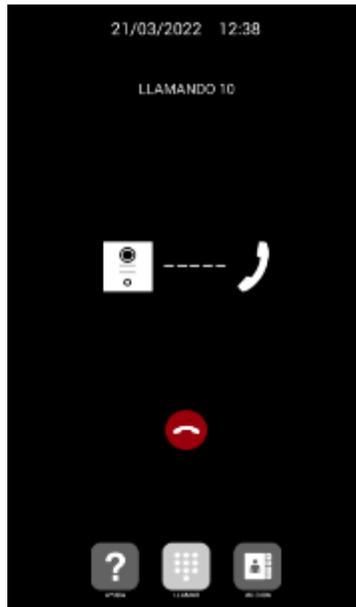


Direct call by code.



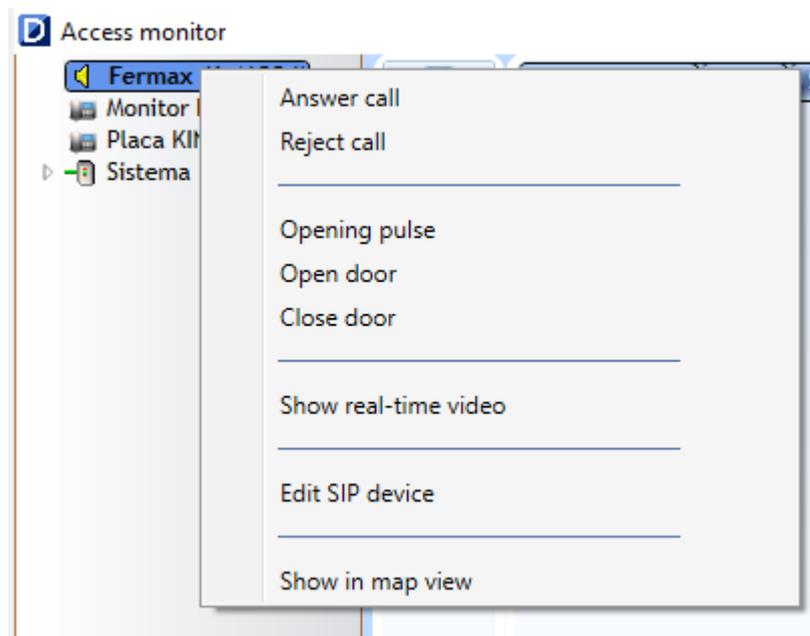
Call via phonebook.

The panel will generate the call as if it were a house:

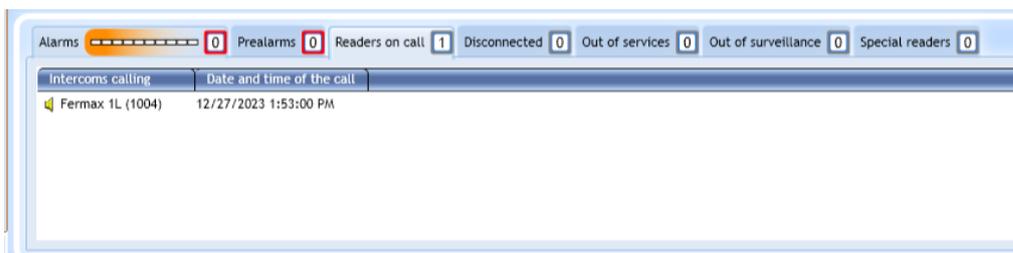


The called extension will indicate the origin of the call by displaying a speaker icon next to the name assigned in the PBX and the chosen sound will be played. You have a maximum of 30 seconds to accept the call.

The call is represented in the access monitor as follows:



In the architecture tree the icon changes when we receive the call and in the lower area, in readers, the call also appears in the information.



If we right click on the name, a menu will appear where we can answer or reject the call.

When the call is accepted, two-way audio communication shall be established for up to 120 seconds and the image from the outdoor panel camera shall be displayed.

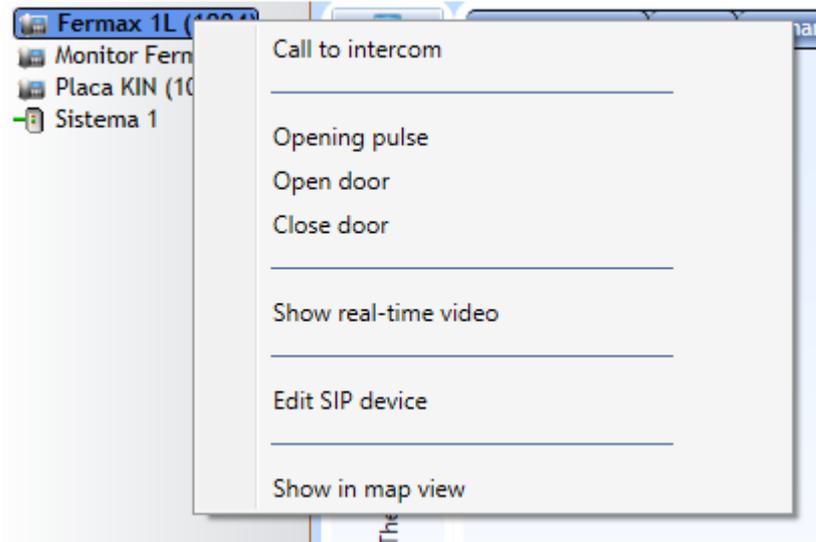
Once the call is established, we observe that the video and several options are displayed, since we have assigned the SIP device to a reader.



- **Swipe card**: This option opens the menu to select a card stored in the database.
- **Opening impulse**: We perform an opening impulse on the access.
- **Open door**: We perform the action of leaving the door permanently open.
- **Close door**: We perform the action of closing the door.

#### Auto-on:

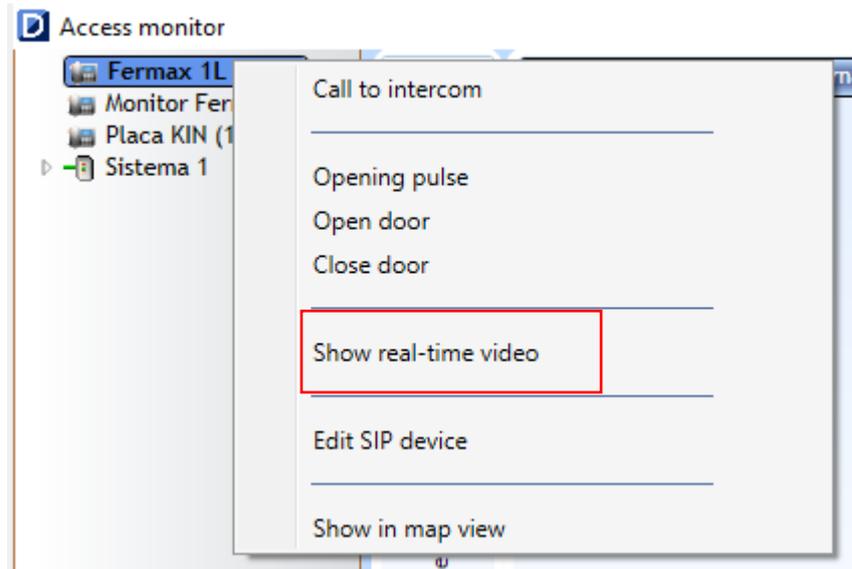
From DASSnet® we can make calls to the intercom by selecting it with the right mouse button and choosing the call option.



You will have the same options as for incoming calls.



From the previous menu we also have the option to view the camera in real time without having to make the call.



## OTHER CONSIDERATIONS

### Compatibility with MEET monitor:

If a MEET monitor is available, the call can be received simultaneously at the called extension and at the monitor. The monitor must be assigned the call code dialed on the outdoor panel as the house number. The first one to answer the call will cut off the reception at the other one.

Another option is to configure the monitor in SIP mode as another extension of the switchboard, allowing it to be called from any extension or to call these extensions from the monitor by means of the 'Extercom' option, entering the extension number. **In this mode, the video preview is lost until the call is answered when picking up.**

For this mode, a new extension must be registered in the SIP PBX with the same configuration that was given to the outdoor panel.

Next, to configure the monitor, access the monitor's web server from a browser using its IP address, and configure the credentials of this extension in the SIP tab:

The screenshot shows the FERMAX web interface for the MEET VIDEO DOOR ENTRY SYSTEM. The left sidebar contains a menu with options: DEVICE, GENERAL, NETWORK, IP CAMERA, SIP (selected), ADVANCED, ACTUATORS, VERIFICATION, PINCODE, and RESTORE. The main content area is titled 'SIP SETTINGS' and includes the following fields:

- ENABLE SIP:  [SEARCH SIP STATUS](#)
- SIP SERVER: sip:192.168.1.191
- DOMAIN: 192.168.1.191
- OUTBOUND: [Empty field]
- STUN IP: [Empty field]
- STUN PORT: 5060
- SIP USER: 6010
- SIP PASS: [Masked with asterisks]
- CONVERSATION: 120S

A 'SAVE' button is located at the bottom of the configuration area.

In the ADVANCED option, an opening DTMF tone must be configured in case you want to activate the outdoor panel door release.

The screenshot shows the FERMAX web interface for the MEET VIDEO DOOR ENTRY SYSTEM. The left sidebar contains a menu with options: DEVICE, GENERAL, NETWORK, IP CAMERA, SIP, ADVANCED (selected), ACTUATORS, VERIFICATION, PINCODE, and RESTORE. The main content area is titled 'ADVANCED SETTINGS' and includes the following fields:

- SIP EXT.: DISABLE
- AUTO ANSWER:
- ONU(GPON):
- DTMF UNLOCK:
- DTMF KEY: #
- NUMBER OF DOORLOCKS: 0
- NUMBER OF CAMERAS: 0

A 'SAVE' button is located at the bottom of the configuration area.

Call forwarding to MEETME application:

The outdoor panel can be configured to make the call to the MeetMe application simultaneously or sequentially, with the advantage of being able to receive it wherever the operator is located. It is necessary to purchase a licence for the Meetme call service ref. 1496.

In the configuration of the calling extension to which the outdoor panel is to call, the extension of the SIP client shall be indicated and then, separated by ';' the extension of the licence.

APARTMENT,NUMBER,			
1,sip:6008@192.168.1.191			
5,sip:6009@192.168.1.191			
3,sip:6010@192.168.1.191,			
4,sip:6011@192.168.1.191;sip:00879892@sip.fermax.com,			

In this example, dialing call code 4 will call extension 6011 and also the MeetMe app that has been registered with the licence 00879892.

The parallel or sequential mode is configured on the outdoor panel:



DEVICE	GENERAL SETTINGS		
GENERAL	TYPE:	1W PANEL	
NETWORK	BLOCK:	1	
ACC	APARTMENT:	101	
SIP	DEVICE NO.:	1	
SIP TRUNK	DEVICE TAG:	FERMAX (≤16 CHARACTERS)	
SIP CALL	LANGUAGE:	ENGLISH	
ADVANCED	PANEL VOLUME:	5	
PINCODE	DOOR OPEN VOICE:	<input checked="" type="checkbox"/>	
RESTORE	VIDEO RESOLUTION:	640x480	
	SIP DIVERT MODE:	PARALLEL CALL	
	DATE FORMAT:	DD/MM/YYYY	
	DATE:	01 / 01 / 2018	
	TIME:	20 : 23 : 55	
	TIME ZONE:	GMT+01:00	
	SAVE		