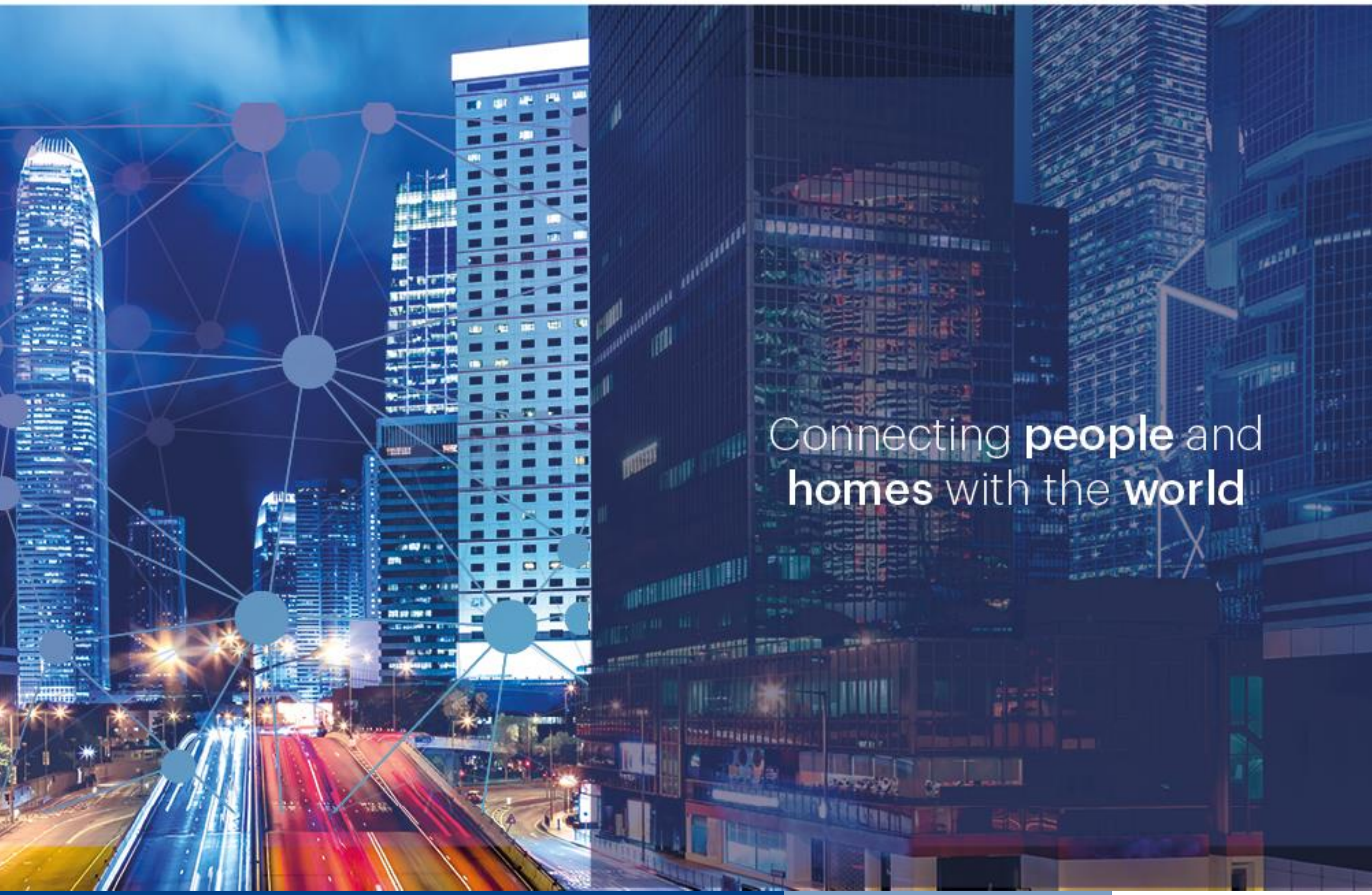


MARINE SIP PANEL INSTALLER MANUAL

DATE:

February 2025

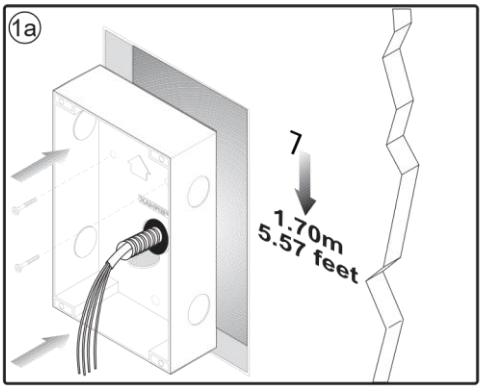
Cod. 970300Ic

INDEX

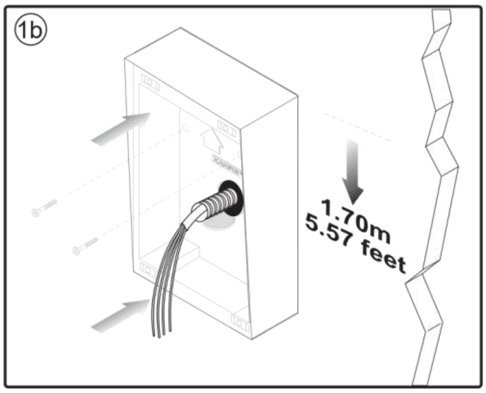
1 FIRST STEPS

INSTALLATION

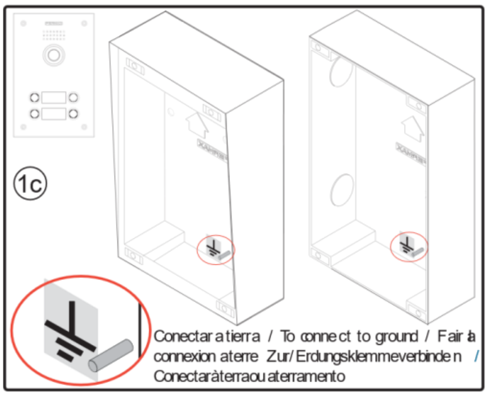
POWER SUPPLY INSTALLATION

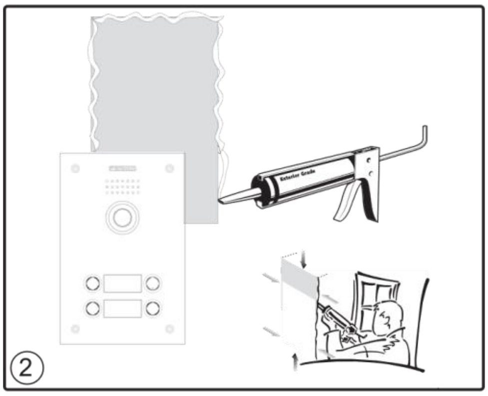


- Installation with FLUSH-MOUNTED BOX

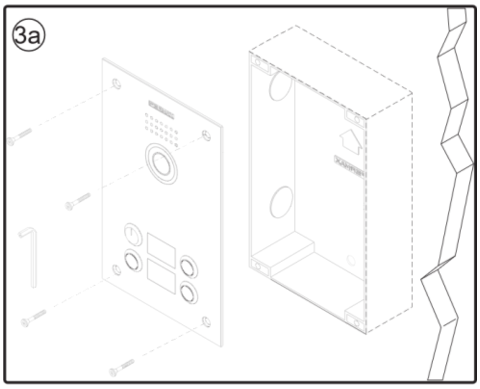


- Installation with SURFACE-MOUNTED BOX

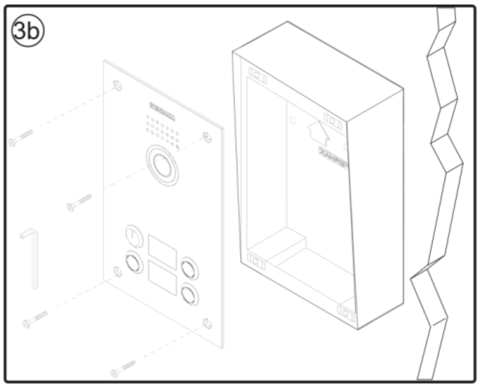




We advise you to seal the panel with silicon to ensure watertightness.



Cover for panel with FLUSH-MOUNTED BOX



Cover for panel with SURFACE-MOUNTED BOX

Default configuration

User & password	Value	*Webserver*
User	admin	-
Password	123456	System » Account » User Management

Network configuration		Network » Basic » IPv4 Settings » …
Type	Static IP	
IP address	10.1.0.1	
Mask	255.0.0.0	
Gateway	10.254.0.1	

Push buttons	*Action executed when a certain button is pressed (short press)*	
(Bottom) **1**	10.1.1.1 (Type *Meet*)	Function Key » Function Key Settings » DSS Key 1
2	10.1.1.2	
3	10.1.1.3	
4	10.1.1.4	
(Top) **5**	10.1.1.5 (Type *Meet*)	… » DSS Key 5

Tags	*Names displayed when calling (via P2P or SIP server) to a certain device*	
MEET devices (P2P)	PANEL	Line » Basic Settings » SIP P2P Settings » User name
MEET ME app (SIP)	MEETME	Line » SIP » Register Settings » Display name

SIP servers	*Local or remote SIP server, if necessary. Domain name or IP address*	
SIP server 1 (MEET ME@SIP1)	sip.fermax.com	Line » SIP » SIP Server *1* » Server Address
SIP server 2 (SIP2)	-	Line » SIP » SIP Server *2* » Server Address

Output		Security Settings » Output Settings » …
Idle status	C-NC closed*	» Standard status
Duration	4 seconds	» Output Duration
DTMF tone	#	» DTMF Trigger Code
Feedback tone	bell.wav	» Triggered by DTMF Ring Tone

*Refers to the status of the relay in passive -non-powered- state. See serigraphy on the back of the device

Important: the door lock must be supplied via an external PSU.

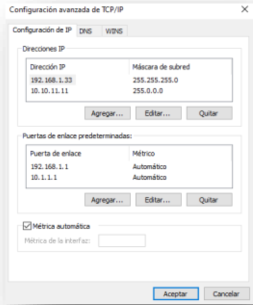
Maximum capacity Output1: 2A@30VDC, 0.5A@125VDC

*Each push button accepts **up to 8 different values** (IP addresses or MEET ME licences) only if Type = MEET. For any other type, only 1 value is allowed.*

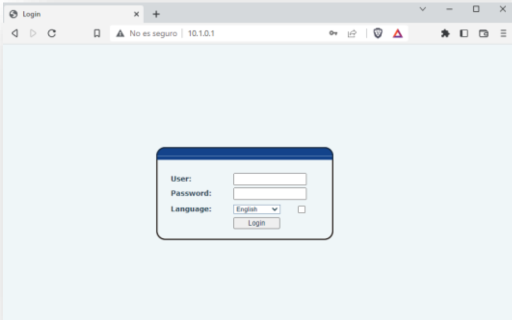
Changing MARINE SIP panel default IP address

1. Add the following configuration to your PC's network interface adapter:

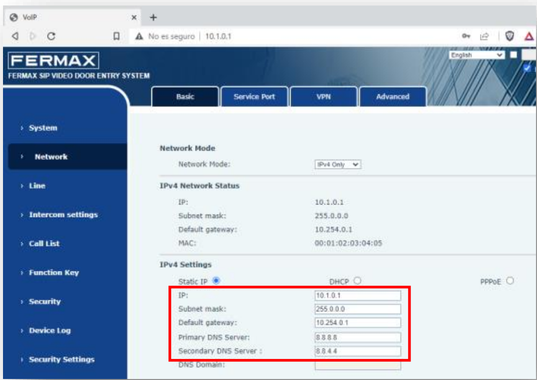
   IP address: 10.10.11.11
   Mask: 255.0.0.0



2. Open your **web browser**, navigate to http://**10.1.0.1** and set its default **user and password**:



3. Go to **Network** menu and change default **IPv4 Settings** according to the configuration of your local network.



In case you don't know the IP address of the panel:

1. *Press and hold push button #1 (bottom). After some seconds, a sound will be emitted.*
2. *Immediately, press push button #1 shortly. The panel will announce its current IP address.*

Other parameters

Settings	Webserver
Audio	Intercom settings » Media Settings » … • Codecs • Volume
Video	Intercom settings » Camera Settings » … • Codecs • Resolution, Frame rate…
SIP P2P	Line » Basic Settings
SIP servers (1/2)	Line » SIP » … • Transport protocol • Port

2 MEET INSTALLATIONS

This chapter is intended to summarize the operation and particularities of MARINE SIP panel when installed in **MEET 3.50 installations**: local SIP P2P call to MEET devices and/or remote call to MEET ME app using FERMAX cloud SIP server.

It is possible to mix MEET panels (KIN, MILO,…) without external relay modules ref. 1490 or 1491 or lift control and MARINE SIP panels within the same installation, calling to the same MEET devices and/or smartphones with MEET ME app installed.

It is possible to mix Meet Guard Unit ref. 95391 and MARINE SIP panels within the same installation but Image capture and video recording functions are not supported.

Please consider the following table as a reference for knowing the constraints of this integration:

MEET device (min. v3.50)	**Available**	**Not available**
All MEET devices	Audio, Video & Door opening Identification tag (default = PANEL) Call log Video preview Sequential call to multiple MEET devices	Simultaneous call to multiple MEET devices via P2P call (possible with a SIP server) Panel Auto ON
NEO monitor		Call back via Call log
WIT A10 monitor	Manual picture capture Panel Auto ON (via Trigger APP)	
Guard unit	Call back via Call log	Door opening via Call back function Modes (Day, Night, Mixed) Call Forward & Call Transfer Manual & Automatic picture capture

MEET ME app	**Available**	**Not available**
iOS/Android	Audio, Video & Door opening [2] Identification tag (default = MEET ME) Call log & Call back via Call log Sequential call to multiple MEET ME lic. Simultaneous call to up to 8 smartphones using the same licence [1] Panel Auto ON (once the app receives a first call from the panel) Call reception while APP is closed	External cameras Auxiliary relays
Android	Video preview in multiple devices	
iOS		Video preview (due to iOS restriction)

Incompatibility with the Lift control function, Marine SIP panel and guard unit in the same installation. Video capture and recording functions from the panel are not compatible if there is a guard unit.

[1] This particular scenario refers to a MARINE SIP panel with a MEET ME license calling to a single MEET ME license, shared by up to 8 different Android/iOS smartphones. In this situation, attending or rejecting an incoming call from any smartphone automatically hangs on the call in the rest of smartphones.

Call from MARINE SIP Panel to MEET ME licences requires to purchase **1x ref. 1496 / MEET ME LICENCE per each SIP panel**, plus a certain number of additional ref.1496 which will depend on the number of independent smartphones to be called. All MEET devices include 1x ref. 1496 which can be used for this purpose.

[2] iOS devices: Door opening function may require waiting from 6 to 8 sec. once the call is answered to ensure that the action is executed correctly.

It is possible to **call sequentially** -call to device 1, no answer, call to device 2,…- to multiple devices (MEET monitors, MEET ME licences…). Smartphones sharing the same licence will ring at the same time. See example ***M3****. Call to multiple devices*. Parallel call -call to devices 1 and 2 at the same time- to several MEET devices requires of a dedicated SIP server, and additional configuration.

Sumary of examples:

- **M1.** MARINE SIP panel CP101. Local P2P call to a MEET monitor.
- **M2.** MARINE SIP panel CP101. Remote call to a MEET ME licence.
- **M3.** MARINE SIP panel CP105. Local and Remote call to 5x WIT A10 monitors + MEET ME licences.

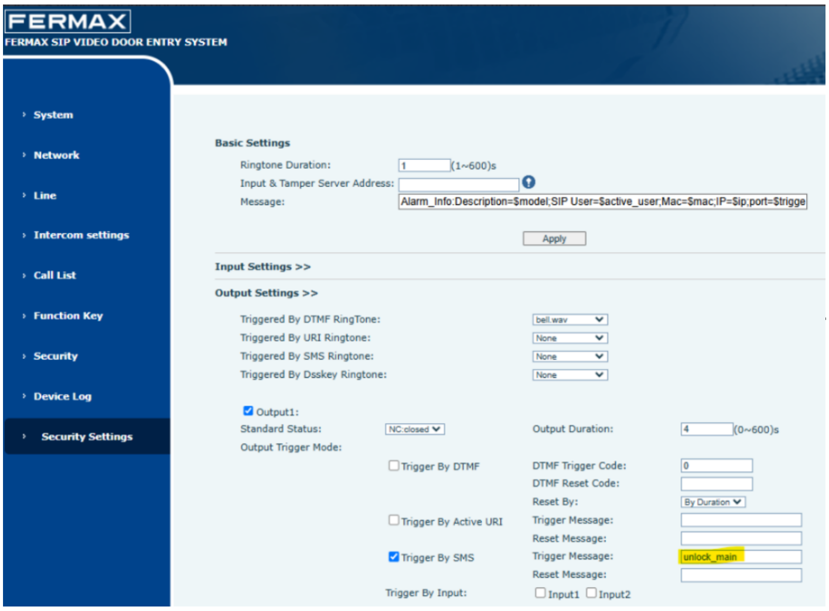
These three examples can serve as the basis for any other possible scenarios related to installations with MEET devices.

Door opening with the new MeetMe app

Opening the front door

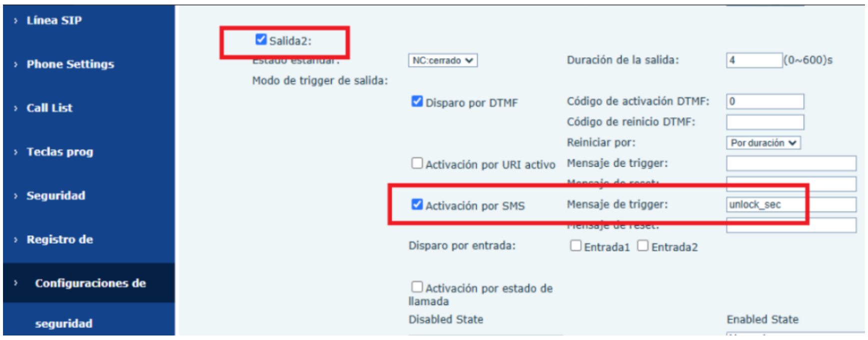
In order to open the front door with the new app, the S**ecurity Settings > Output Settings > Select the desired output (Output1/Output2) > Trigger by SMS > unlock_main must be activated.**

From firmware 5.8 it is activated by default. The firmware file can be downloaded from the product sheet on the fermax.com website.



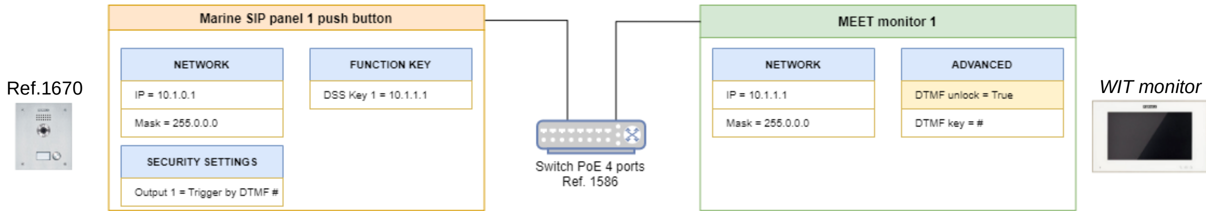
Secondary door opening

Additionally, if we want to control the second SIP board relay, we will have to repeat the steps for Output 2, using in this case the SIP message **'unlock_sec'.**



<u>Examples</u>

M1. 1x Marine SIP panel CP101 (1 push button) + 1x WIT monitor (P2P call). Local-only installation.

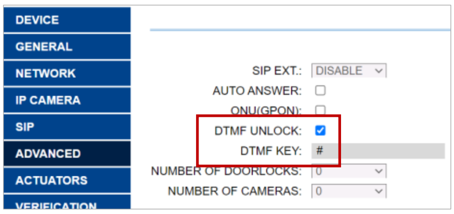


Background in white = default configuration; yellow = changes to apply

Steps:

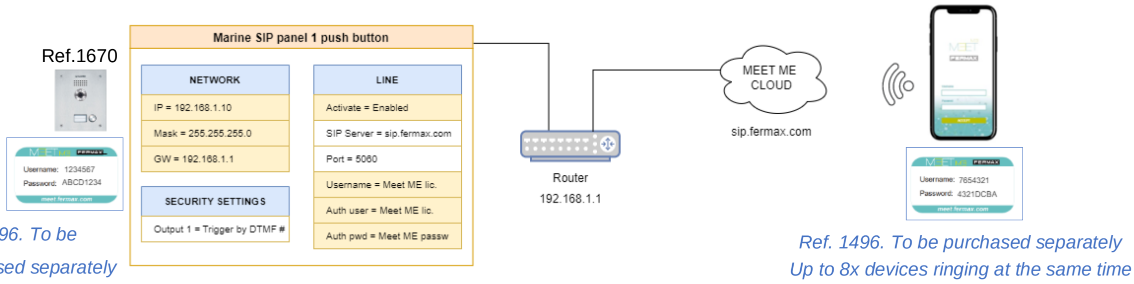
1. Network: no additional configuration required.
2. Call panel to monitor: no additional configuration required.
3. **MEET monitor**: DTMF command for Door opening.

   3.1. Log into the monitor (see its corresponding manual for additional details).
   3.2. Go to the **Advanced** menu and enable **DTMF Unlock** option. Ensure **DTMF Key** = #



Once saved, the monitor will send the DTMF tone “#” <u>while in conversation</u> with the panel.

M2. 1x Marine SIP panel CP101 (1 push button) + 1x smartphone with MEET ME app + licence ref. 1496 (remote SIP server, MEET ME cloud). Remote-only installation (internet connection required).



Background in white = default configuration; yellow = changes to apply

Steps:

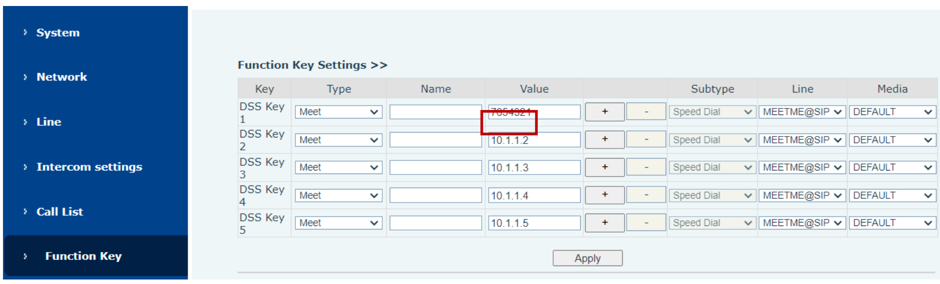
1. Network: change the configuration of the router and/or Marine SIP panel (see 1. First Steps)

2. **MARINE SIP Panel**. Call to MEET ME:
   2.1. Webserver: Go to **Line** menu. Line: MEETME@SIP1.
   2.2. Register the device on **sip.fermax.com** using the purchased MEET ME license
      1) Check **Activate** option.
      2) Set **Username** & **Authentication User** to MEET ME Username (i.e.: 1234567).
      3) Set **Authentication Password** to MEET ME Password (i.e.: ABCD1234).
      4) Optional: Change **Display name**. Name displayed in MEET ME App when the device calls to any MEET ME license. Default: "MEETME".
      5) Select the desired communication protocol and port
         - With UDP, the port is the same (5060) and you just have to reboot.
         - With TCP (default option), the port has to be changed to 5223.
         - With TLS, the port must be changed to 4443 (recommended)..
      6) Apply changes. Result: **Line Status = Registered.**

Other possible *Line Status* (5) messages:

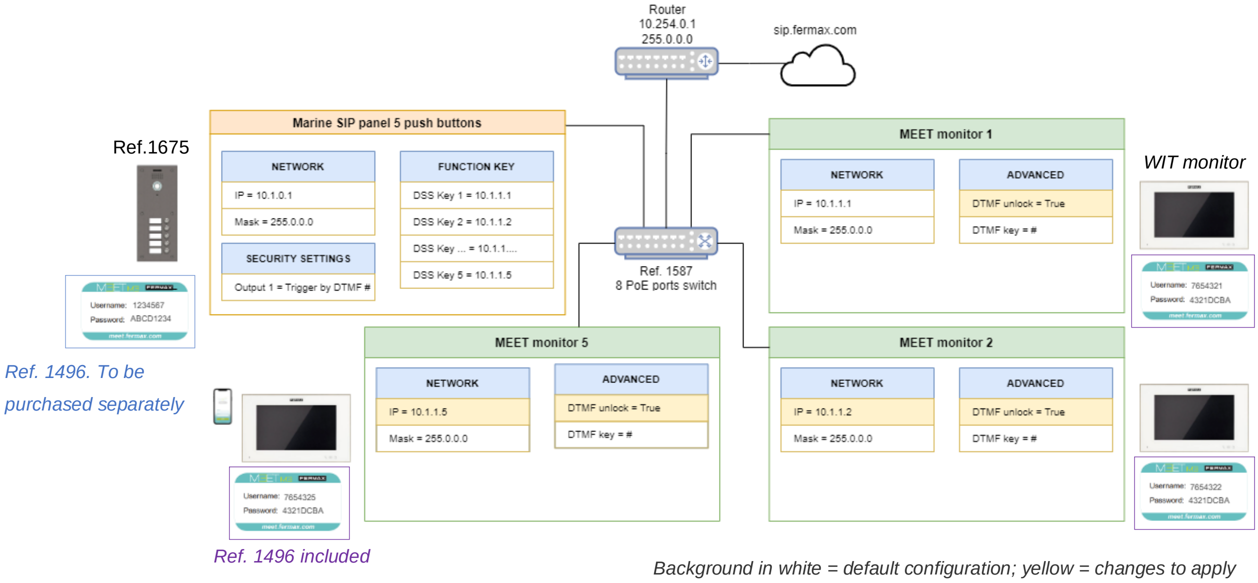
- **Timeout:** no answer from server. Check Network parameters & SIP Server 1 configuration.
- **Failed with 403**: answer from server, but connection rejected. Check Username, Authentication User & Authentication Password fields.

   2.3. Go to **Function Key** menu. Set the number of MEET ME license to be called when the button is pressed.



2.4. Door opening configuration.

M3. Local + Remote call. 1x Marine SIP Panel CP105 (5 push buttons) + 5x WIT monitors. Each monitor includes 1x MEET ME licence (local P2P call + remote SIP server, MEET ME cloud)



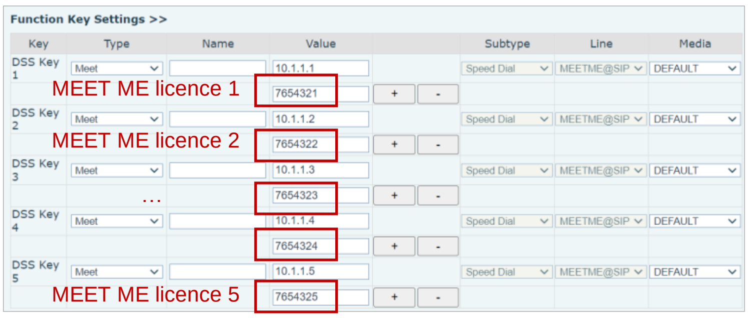
Background in white = default configuration; yellow = changes to apply

Steps:

<u>Marine SIP Panel</u>

1. See example ***M2***, step 2 (MARINE SIP PANEL, call to MEET ME), for enabling “call to MEET ME devices” functionality.
2. Go to **Function Key** menu. Default **Function Key Settings**:

3. Call to MEET ME licences: For each DSS Key, press “**+**” button and add the number of each MEET ME licence included on the front of each MEET monitor



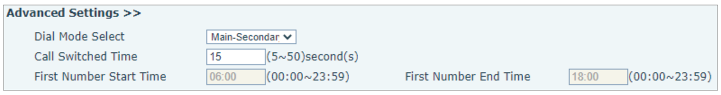
Function Key Settings >>

Key	Type	Name	Value			Subtype	Line	Media
DSS Key 1	Meet		10.1.1.1			Speed Dial	MEETME@SIP	DEFAULT
		MEET ME licence 1	7654321	+	-			
DSS Key 2	Meet		10.1.1.2			Speed Dial	MEETME@SIP	DEFAULT
		MEET ME licence 2	7654322	+	-			
DSS Key 3	Meet		10.1.1.3			Speed Dial	MEETME@SIP	DEFAULT
		…	7654323	+	-			
DSS Key 4	Meet		10.1.1.4			Speed Dial	MEETME@SIP	DEFAULT
			7654324	+	-			
DSS Key 5	Meet		10.1.1.5			Speed Dial	MEETME@SIP	DEFAULT
		MEET ME licence 5	7654325	+	-			

Once changes are saved, if any push button is pressed:

1. The panel will call to MEET monitor 10.1.1.X *(Main value)*
2. In case there is no answer after 15 seconds, the panel will automatically end the call with MEET monitor and will call to MEET ME licence 765432X *(Secondary Value)*

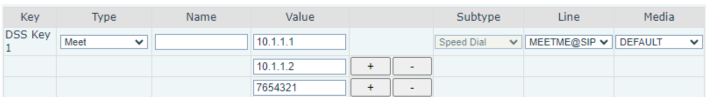
Default behaviour can be modified for all push buttons via **Function Key** menu, **Advanced Settings**:



Dial mode	Options: • Main-Secondary: call to Main. If no answer after *Call Switched time*, then call to Secondary • Time Period: call to “Main” during the period between Start-End time. If out of this period, then call to “Secondary” Default: Main-Secondary
Call Switched time	Minimum: 5 seconds Maximum: 50 seconds (30 seconds in MEET installations) Default: 15 seconds
First Number Start/End Time	00:00 to 23:59

Each *DSS Key* can include up to 8 different sequential calls to MEET devices or MEET ME licences. Each new added row will act as a “Secondary value” of its previous row.

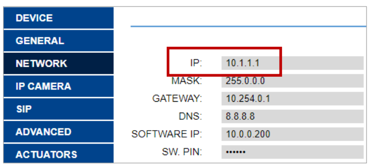
Example: Call to MEET device 10.1.1.1. In case there is no answer, after 15 seconds, call to MEET device 10.1.1.2. In case there is no answer, after 15 seconds, call to MEET ME licence 7654321.



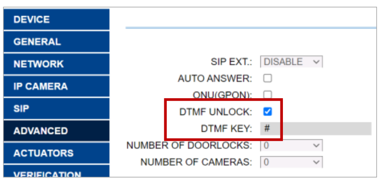
WIT monitors

For each monitor…

1. Change **Network** parameters of monitors 2 to 5: **IP address** 10.1.1.2, 10.1.1.3, etc.



2. Go to **Advanced** menu and enable **DTMF Unlock** option. Ensure **DTMF Key** = #

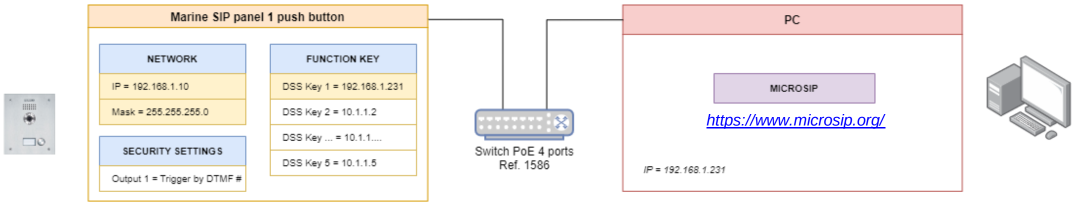


3 SIP INSTALLATIONS

MARINE SIP panel can call to **3rd party SIP devices**, local or remote, via SIP P2P call or SIP server. Below you can find some examples of the configuration of MARINE SIP Panel in each case.

Examples

S1. 1x Marine SIP panel CP101 (1 push button). Local P2P call to a computer with Micro SIP client installed



Steps:

1. Network: change the configuration of the router and/or Marine SIP panel (see 1. First Steps)
2. Go to **Line** menu, **Basic Settings** tab, **SIP P2P Settings**, and set the name to be displayed when the panel calls to the PC. Example:

Display name:	STREET

3. Go to **Function Key** menu and set:
   - Type = Memory key
   - Value = 192.168.1.231
   - Media = DEFAULT (audio + video), alternatives: audio-only, video-only

Function Key Settings >>

Key	Type	Name	Value			Subtype	Line	Media
DSS Key 1	Memory Key		192.168.1.231	+	-	Speed Dial	MEETME@SIP	DEFAULT

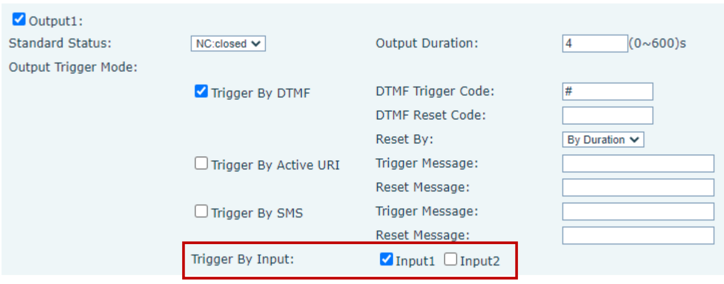
4 HOWTOS

4.1 Inputs & Outputs

- **Inputs I1 & I2** are only available on SIP boards with up to 3 buttons. The 4-button SIP board only has free input I2. No free inputs on 5-button boards.
- **OUT1:** 2A@30Vdc, 0.5A@125Vac
- **OUT2:** Connection to DDA module (Annex I: SIP DDA board) . If there is no module, 2A@30Vdc, 0.5A@125Vac. *See 4.1.2 Out 2 relay configuration.*

4.1.1 Connect an Exit push button

Go to **Security Settings**, **Output Settings** and enable Trigger by Input: Input1 or Input2.

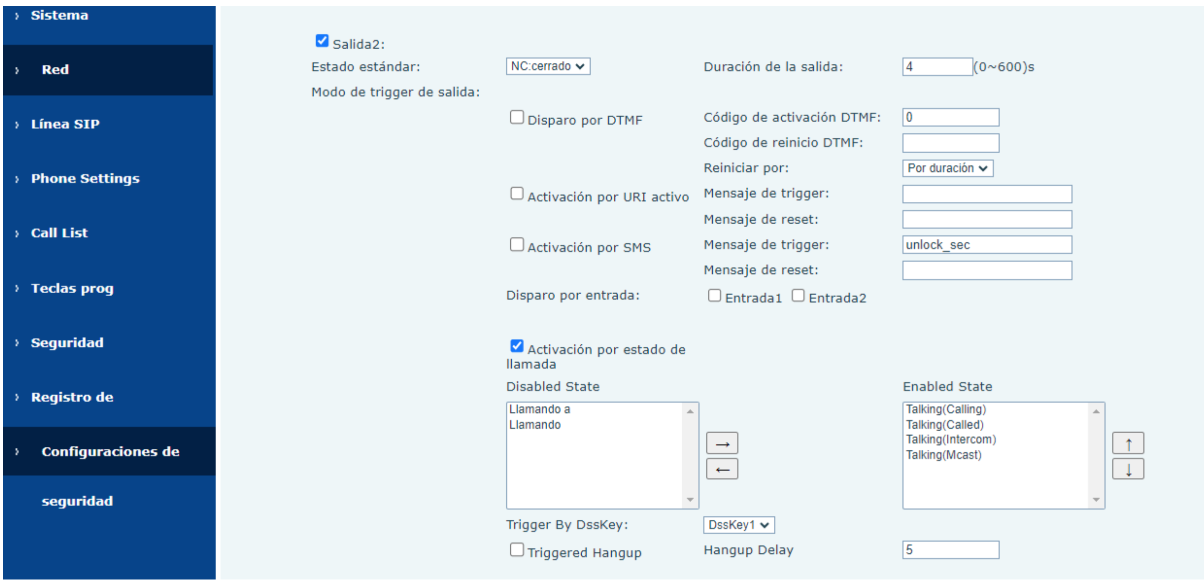


Once enabled, connect the exit push button or relay of proximity reader to configured input.

NOTE: The 'Input parameters' selected in 'security settings' must be configured.

4.1.2 Relay configuration Out 2

La placa Marine SIP dispone de un segundo relé configurable desde el Web server. Si la placa tiene modulo DDA, el módulo está conectado a este relé y por tanto no se podrá usar libremente.



- Security Configurations > Output configurations > Output 2
  - Standard Status: [NC (default) - NO]
  - Output Trigger Mode: Selectable
    - DTMF Trigger
    - Trigger by active URI
    - Trigger by input
    - Trigger by call status
    - Trigger by DssKey

4.2 Integrations

4.2.1 HTTP commands or SIP messages based on events

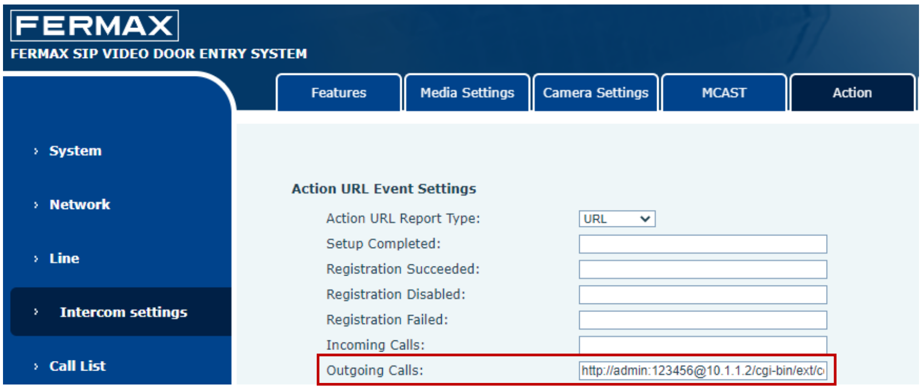
See **Intercom settings** menu, **Action** tab. Main events:

Setup completed	Panel rebooted
Outgoing calls	Call initiated from the panel to any SIP device
Call established	Conversation starts
Call terminated	End of conversation
Output 1	SIP Panel relay activated

All the events will be sent via HTTP (URL) commands or SIP messages. It is not possible to mix different types of actions -some via HTTP command, others using SIP messages- depending on each event.

Example: Send an “Open door” -HTTP command- to a MEET KIN panel when call button is pressed

KIN panel: *http://admin:123456@10.1.1.2/cgi-bin/ext/control.cgi?op=unlock*



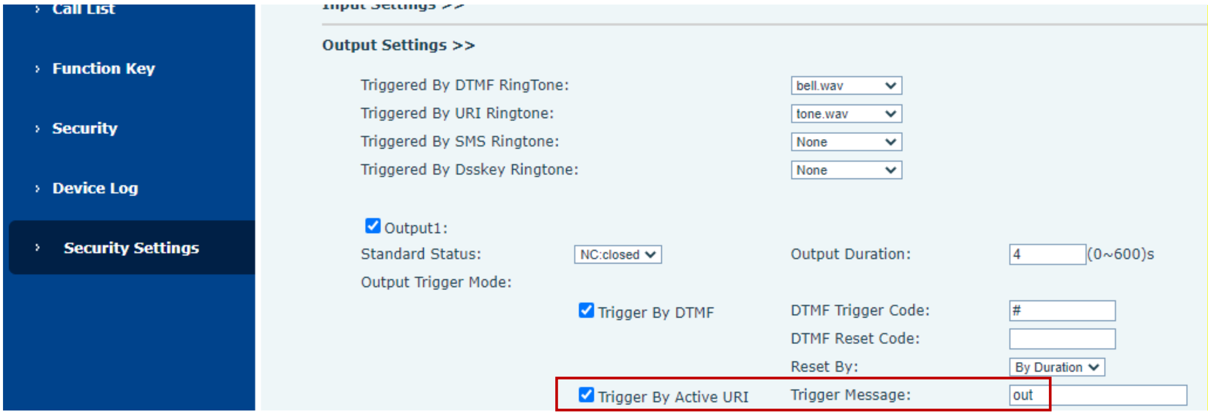
Once the configuration is saved, KIN relay will unlock every time any push button of Marine SIP panel is pressed.

4.2.2 HTTP commands

This functionality allows to remotely control the panel.

Desired action	HTTP command
Press DSS Key 1	http://admin:123456@10.1.0.1/cgi-bin/ConfigManApp.com?key=**DSS1**
Press DSS Key 5	http://.../cgi-bin/ConfigManApp.com?key=
Reboot panel	http://.../cgi-bin/ConfigManApp.com?key=
Trigger Output 1*	http://.../cgi-bin/ConfigManApp.com?

*This action requires additionally to go to **Security menu**, **Output settings**, enable “Trigger by Active URI” and set desired Trigger Message = **out**



The device can send HTTP commands to itself. Example: it is possible to know every time the panel reboots by including a certain MEET ME licence in DSS Key 5 -no initial usage, only for maintenance personnel- which will be called via MEET ME app only when panel reboots:

On Setup completed (panel rebooted) → Remote action = Press DSS Key 5

4.3 Others

4.3.1 Export & Import settings

See **System** menu, **Configurations** tab. These options can be useful to copy the configuration of one device to a different one, and for analyzing the differences between current & default configuration.

From this menu it is also possible to restore the device to its default settings.

4.3.2 RTSP connection

Marine SIP panel has two streams for a continuous connection via Network Video Recorder or similar.

Default URLs:

Main stream	rtsp://admin:123456@10.1.0.1/h264/stream.live0
Sub stream	rtsp://admin:123456@10.1.0.1/h264/stream.live1

A maximum of **3 simultaneous connections** to the panel (main+sub streams) are allowed.

RTSP authentication requirement can be disabled via **Intercom menu**, **Camera settings**, Enable Rtsp Auth (default = enabled).

4.3.3 Taking image captures on events

See **Intercom settings** menu, **Camera Settings** tab.

Events:

Input	Inputs 1 and/or 2 (see IN1 & IN2)
State	Calling, Ringing or Talking
Motion detection	Conversation starts

Actions on event:

A) Save to an SD card (not included)

Steps:

1. Power off the device
2. Insert an SD card (FAT32). See slot on the rear side, top position
3. Power on the device
4. Event simulation: make a call, trigger an input…

All pictures stored on the SD card can be downloaded together -it is not possible to download only a certain picture- using the following command:

http://admin:123456@10.1.0.1/**cgi-bin/cameraPhoto?type=csv**

Filename structure: *call*_year+month+hour+minutes+seconds_*devicecalled*.jpeg

Example: *call_20230517063348_10.1.1.1_5060.jpeg*

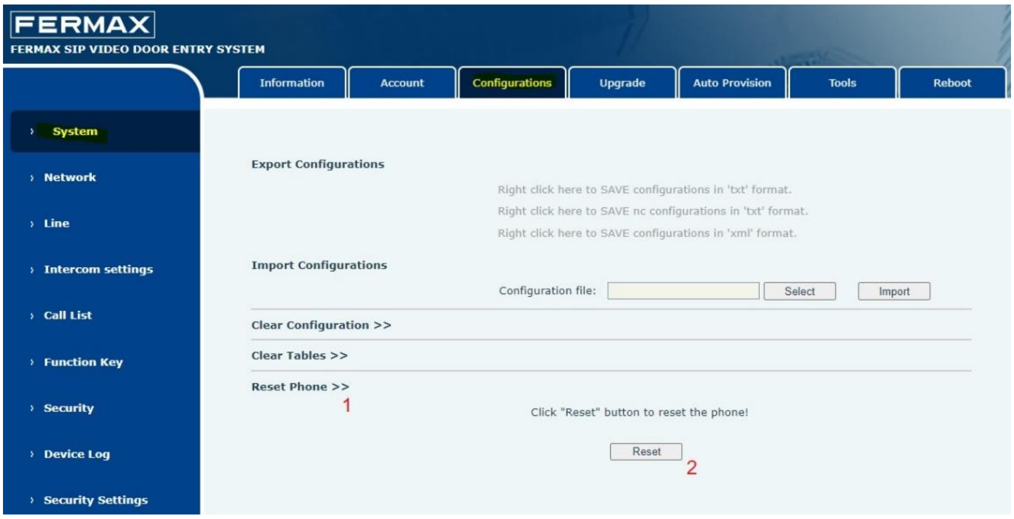
B) Save to an external server (FTP, TFTP, HTTP or HTTPS), using username and password
ftp://server_username:password@IP:port/path

4.3.4 Factory reset

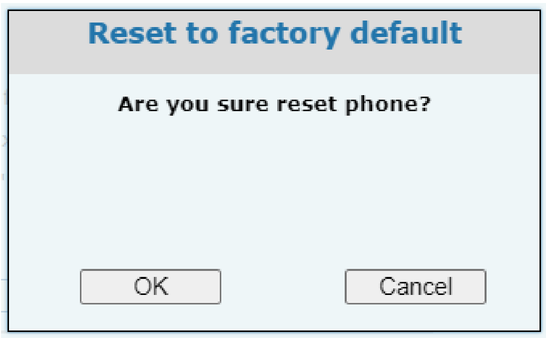
To restore the factory defaults, we can do it from the board's web server or by means of a key combination during boot-up.

From the software

- **System > Settings > Reset phone**



- **Confirm**



From the panel

During start-up (LED crown lit on video panels, call icon lit on DDA or display backlighting on Audio sticks) you can carry out the following sequence on the call button to reset the values to factory defaults:

- 1st Press button number 1.
- 2nd Press push-button number 1 three consecutive times.

The factory settings are detailed in point 1 under ‘Default settings’.

4.3.5 Firmware Update

To enjoy the latest features available, it is recommended to have the newest firmware version installed on the panel.

The firmware is common for all Marine SIP panel references.

Once the panel has been updated, the desired extra functionalities defined in the "changelog" document in the firmware location folder must be configured.

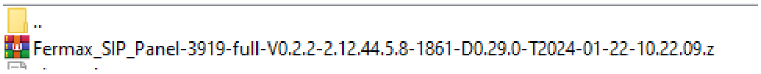
Update:

1. Check the latest firmware available from the technical page of the product in Fermax.com.

   1.1. Download the zipped folder from the “Software” section.

   1.2. Inside the folder is the “changelog.txt” with the new functions and the changes we need to make to activate them.

   1.3. Extract firmware file ‘xxxxx.z’; example:

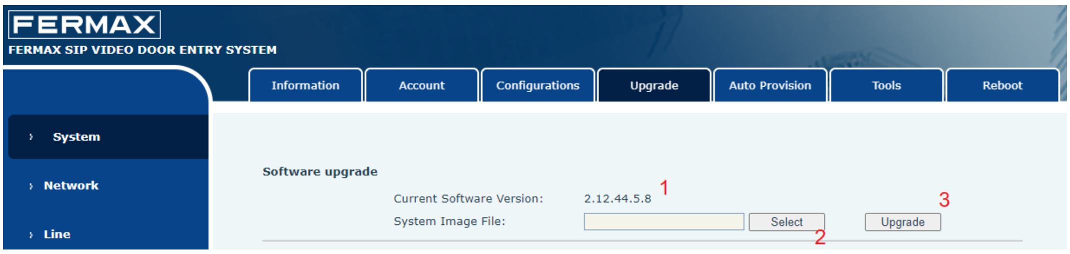


2. Access to the panel's web server.

   2.1. Check the current version of your panel: System > information > software

   2.2. Select the file with the latest firmware: System > Update > ‘select’ and browse to the location of the downloaded file. Select the "xxxx.z".

   2.3. Upgrade; by pressing the pgrade button the process starts. The process will maintain the previous panel c configuration.



Configuration:

Check within the document ‘changelog.txt’ the changes applied to the desired firmware.

Depending on the new function you want, you will have to activate the described parameters.

Example: Activation of ring tone:

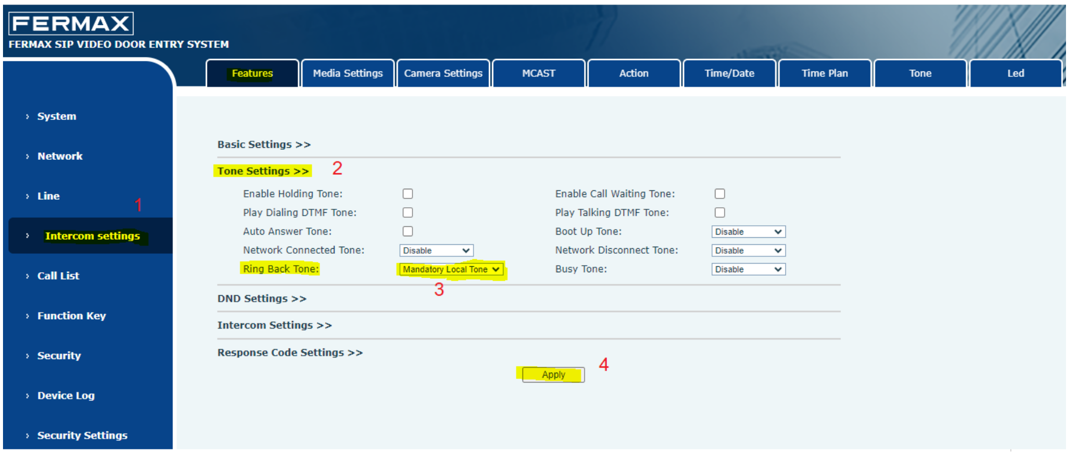
- Changelog’:

-- 2.12.44.5.8

+ Feedback tone

^ See Intercom settings > Features > Tone Settings > Ring Back Tone = Mandatory Local Tone

- Activation:



Note: For upgrades on panels without previous configuration, a "factory reset" is recommended to enable by default all the features added to the new firmware. (See section 4.3.4 Factory Reset).

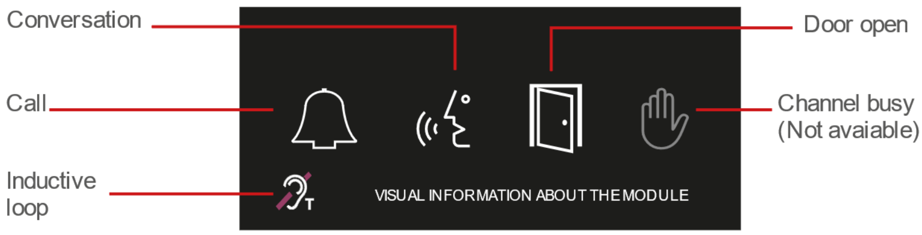
If you have any questions, please contact the technical department.

4.4 Annex I: Panel DDA SIP

To have the One To One function, a module F07452 MODULO ONE TO ONE C/BUCLE IND. V SKYLINE module with special wiring:

The ‘O2O’ modules manage 4 icons:

ICONS



Módulo One to One (DDA) con bucle inductivo

- Call: in parallel with the illumination of the LEDs on the board (video panels). When the board is powered, the call LED is activated for 1 minute.

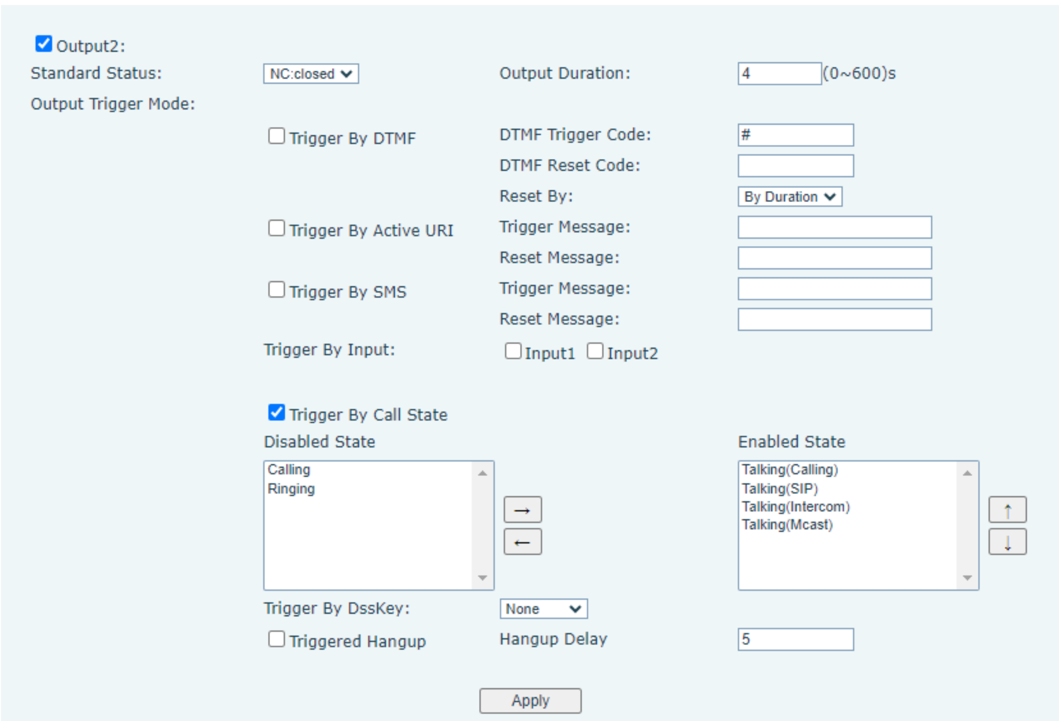
- Conversation: lights up when the call is picked up. Relay Output 2 is used.

- Open door: lights up in parallel with the door release (Output 1).

- Channel busy: not used.

The second relay on the board (Output 2) is configured for use with One to One modules.

Configuration:



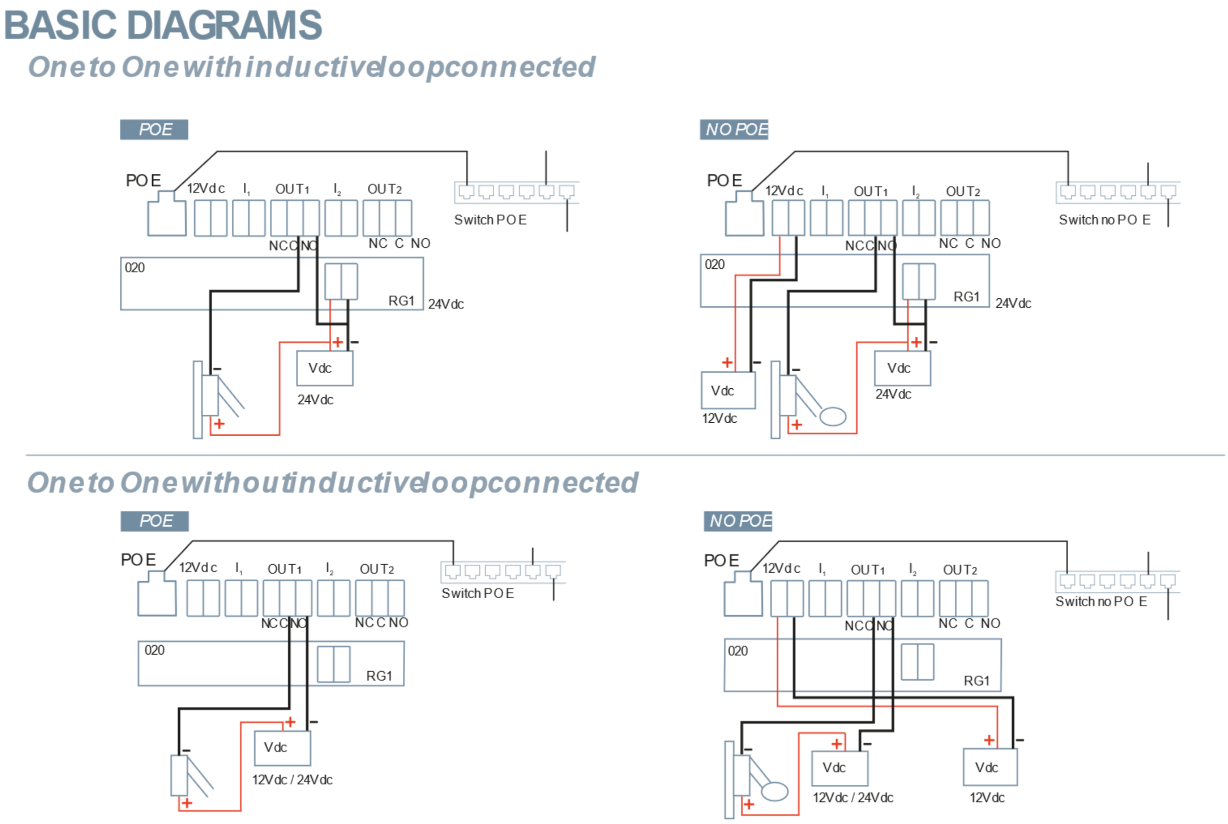
- Security Settings > Output 2.
    - Uncheck Trigger By DTMF.
    - Check Trigger By Call State.

- Leave only the Talking options in Enabled state.

NOTE: The door release must be connected to relay OUT1 on the SIP board. The common (C) of relay 1 is connected to GND.

When using the inductive loop function, it is necessary to use a power supply for the O2O module. The power supply is connected to RG1. This source cannot be shared with the source of the door opener.

Connection:



IMPORTANT INFORMATION ON EQUIPMENT PROTECTION:

Reminder: according to IEC 62368-1, it is necessary to connect the device to a functional earth for protection of the equipment. For this purpose, it is sufficient to connect this functional earth to the flush or surface-mounted box used in the installation of the device.

The device you have purchased is identified under Directive 2012/19/EU on waste electrical and electronic equipment. More info:

Warning:

This device complies with Part 15 of the FCC Rules. Its operation is subject to the following two conditions: (1) this device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

NOTE: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation.

If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

-Reorient or relocate the receiving antenna.

-Increase the separation between the equipment and the receiver.

-Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

-Consult the dealer or an experienced radio technician

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