CASE STUDY

Meet Intercom panel integration with SIP 3CX switchboard

Description

It is described how to configure the Meet entrance panel and the 3CX SIP switchboard to be able to route calls from the entrance panel to any SIP extension.

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V 1.0

INTRODUCTION

In offices and companies, it is common to connect the video door entry panel to the telephone switchboard to receive the call at a specific extension, simplifying the work for employees. For this, a telephone interface is used that connects the entrance panel to an analog input of the switchboard. With the evolution of switchboards, and their deployment in the cloud, a world of previously unimaginable advantages is accessed, such as the reception of calls at any extension located in different places or even on the smartphone, and video reception on the call, which is not available on an analog phone interface. Therefore, it has the same features as in a video door entry monitor if the terminal receiving the call has a screen: two-way conversation, viewing the visit and opening the door.

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

We are going to see, in the specific case of the 3CX SIP switchboard, how the MEET entrance panel and the 3CX switchboard must be configured so that they can work together.

3CX provides 3 types of software PBXs:

- Hosted by 3CX in its cloud

- Hosted in your own cloud (AWS, Google Cloud, MS Azure, …)

- Hosted in the office on a Linux, Windows or Raspberry PI computer.

This document explains how to get up and running with 3CX PBX hosted on its cloud, but the setup is the same in all three cases. 3CX provides a free trial PBX for one year (in any of the modalities) limited to 4 simultaneous calls.

The switchboard supports integration with SIP Trunk providers to make phone calls to non-SIP devices, and integration with Facebook through SIP messaging, to be able to answer messages from the SIP client.

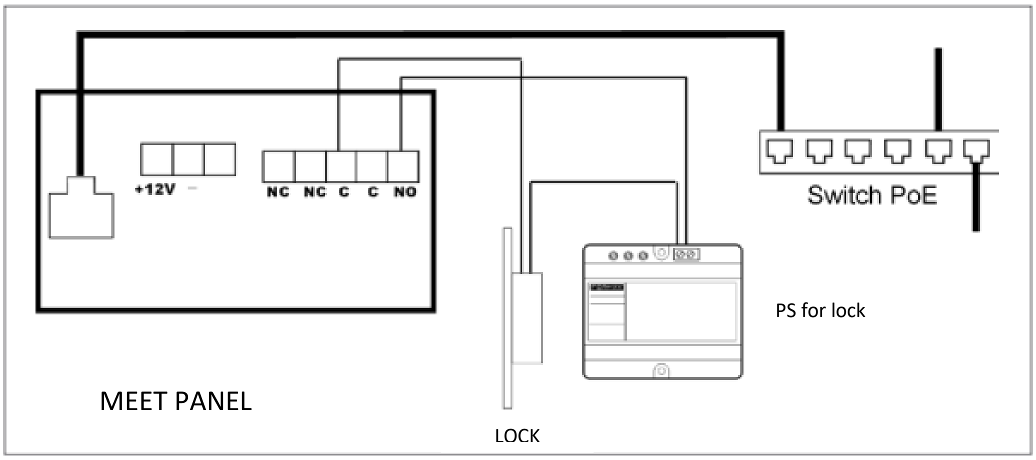
REQUESTED MATERIAL

Any Fermax MEET entrance panel can be used to carry out this integration since they all support SIP protocol. In the case of calling one single extension, it is advisable to use the one-line MILO panel (a single push-button). 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.

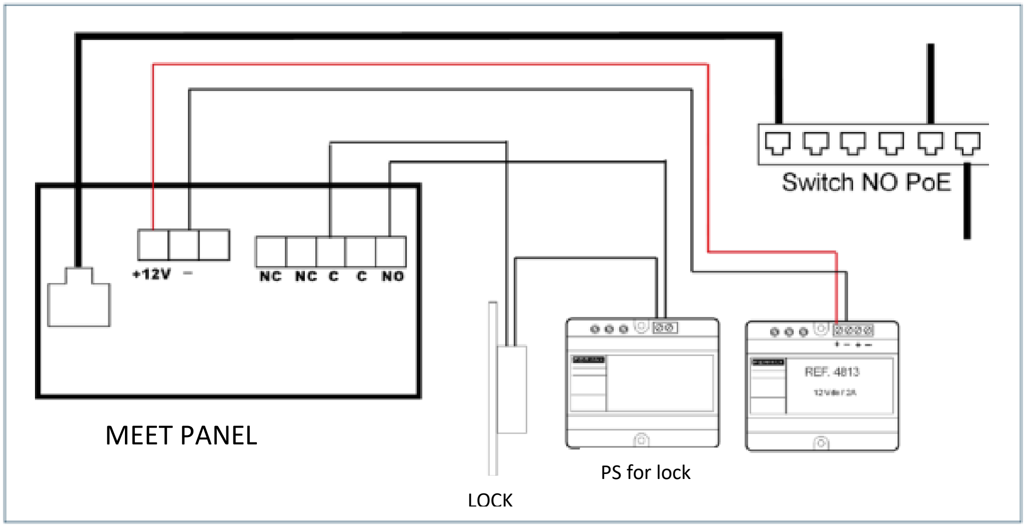
The entrance panel may require a power supply if it does not support PoE, in addition to requiring the electric door opener.

The 3CX switchboard is normally used in cloud format, so only an internet connection with sufficient bandwidth (minimum 10Mbit) will be necessary to access it.

BASIC SCHEMA



Panel power supply via PoE switch.



Panel power supply by additional source. The power supply can be shared with the lock release if both are 12 Vdc voltage and has sufficient power.

INSTALATION

Once the entrance panel is installed, it must be connected to a switch or directly to the installation router. The lock release connection will be wired to the entrance panel or, if maximum security is required, a reference 1490 module with 2 relays will be used to open from indoors, wiring the module to the panel using 3 wires (Cat-5 recommended). Lastly, the panel will be wired to the power supply.

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

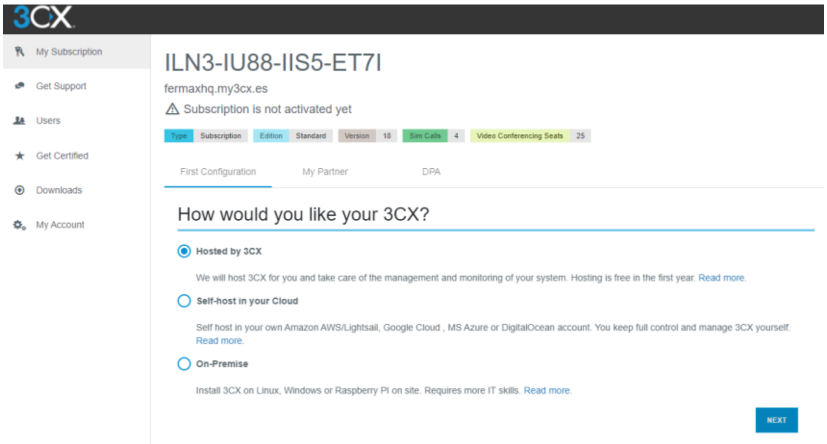
CONFIGURATION

<u>3CX SWITHBOARD</u>

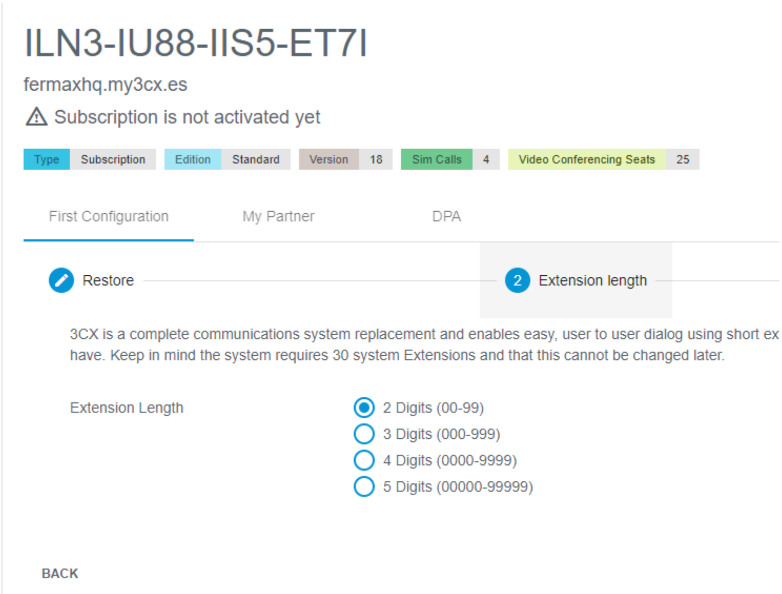
If you do not have a switchboard, you must create a new one. The process is described below. If it is already available, skip this step.

<u>3CX PBX creation in the cloud.</u>

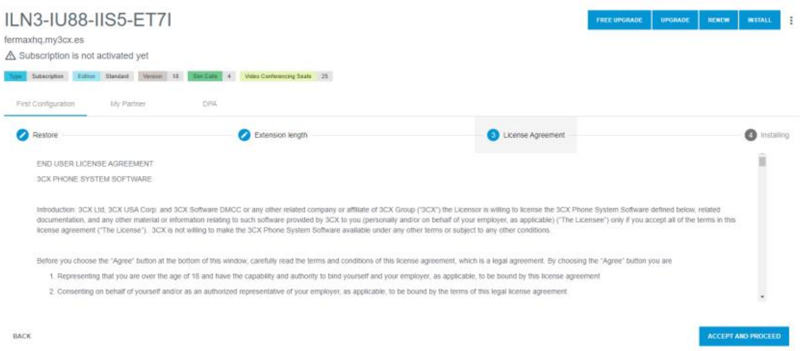
- The web https://www.3cx.es/ is accessed to create the switchboard, in our test case for one year.
- The 3CX hosted option is chosen:



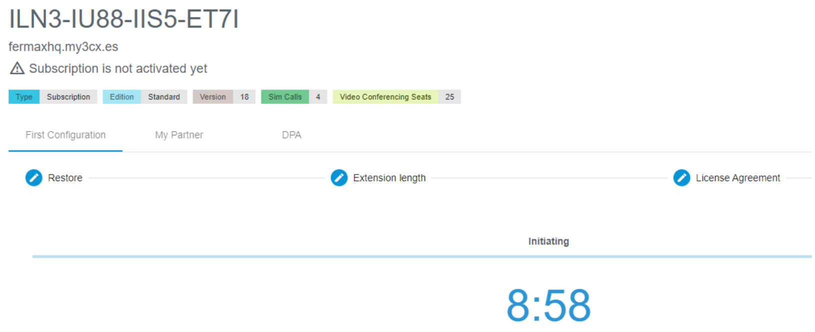
- It will assign a domain name by default, but it can be customized.
- Choose number of digits for extensions: 2, 3, 4.5:



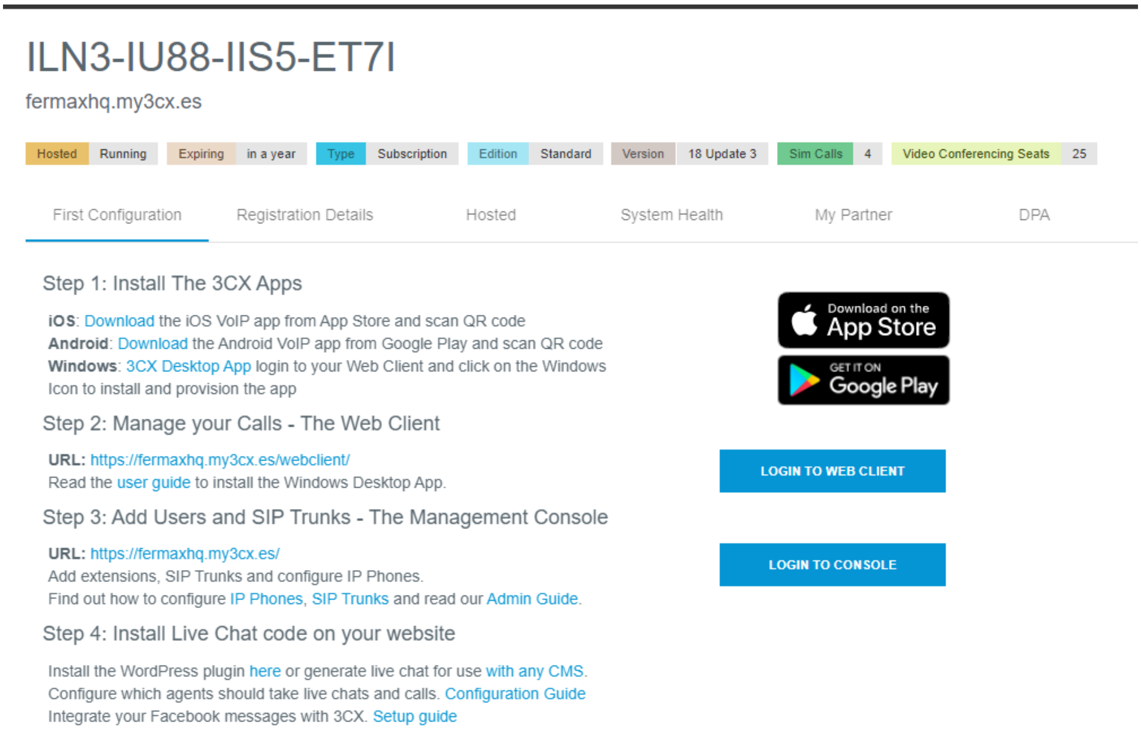
- Accept terms and conditions.



- Start the installation with a 9-minute countdown:



- Installation is finished:



3CX does not have SIP phones but does provide a large list of compatible third-party devices (Fanvil, snom, Yealink, Grandstream) and different SIP software clients to integrate with your PBX:

- Web Client. It is a web page where you can receive and send SIP audio and video calls. It is very practical because any browser can be used.
- Apps. It provides apps for iOS and Android. In this case, the video does not support the H264 codec used by MEET, so it is not compatible with Fermax.
- Windows client. It is an application that is installed on a PC, similar to the web client.

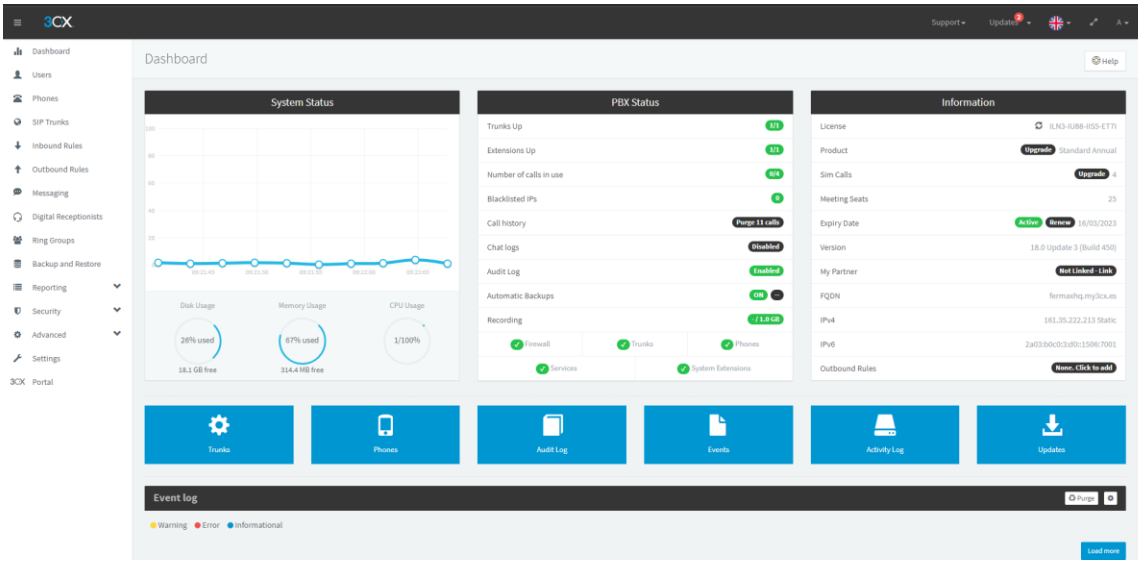
Configuration of extensions in switchboard.

At the end of the switchboard installation, an email is received indicating that a first extension (10) has been created. To test it, there are 2 options:

- Download the app and scan the QR code included in the email to facilitate the configuration of the App.
- Use the web client, a link and the username and password are attached.

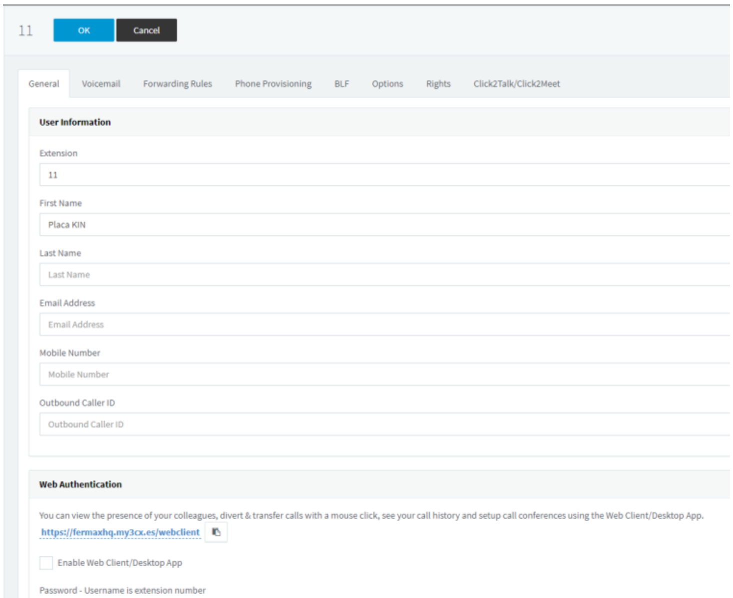
A second email is received to access the web console to manage the extensions. It includes a link to the console, a username and password. In the email you will receive another link to manage the 3CX account created and be able to renew or change the subscription.

The web console is accessed to register an extension for the MEET entrance panel:



The first thing that is shown in the console is a Dashboard with the status of our SIP PBX.

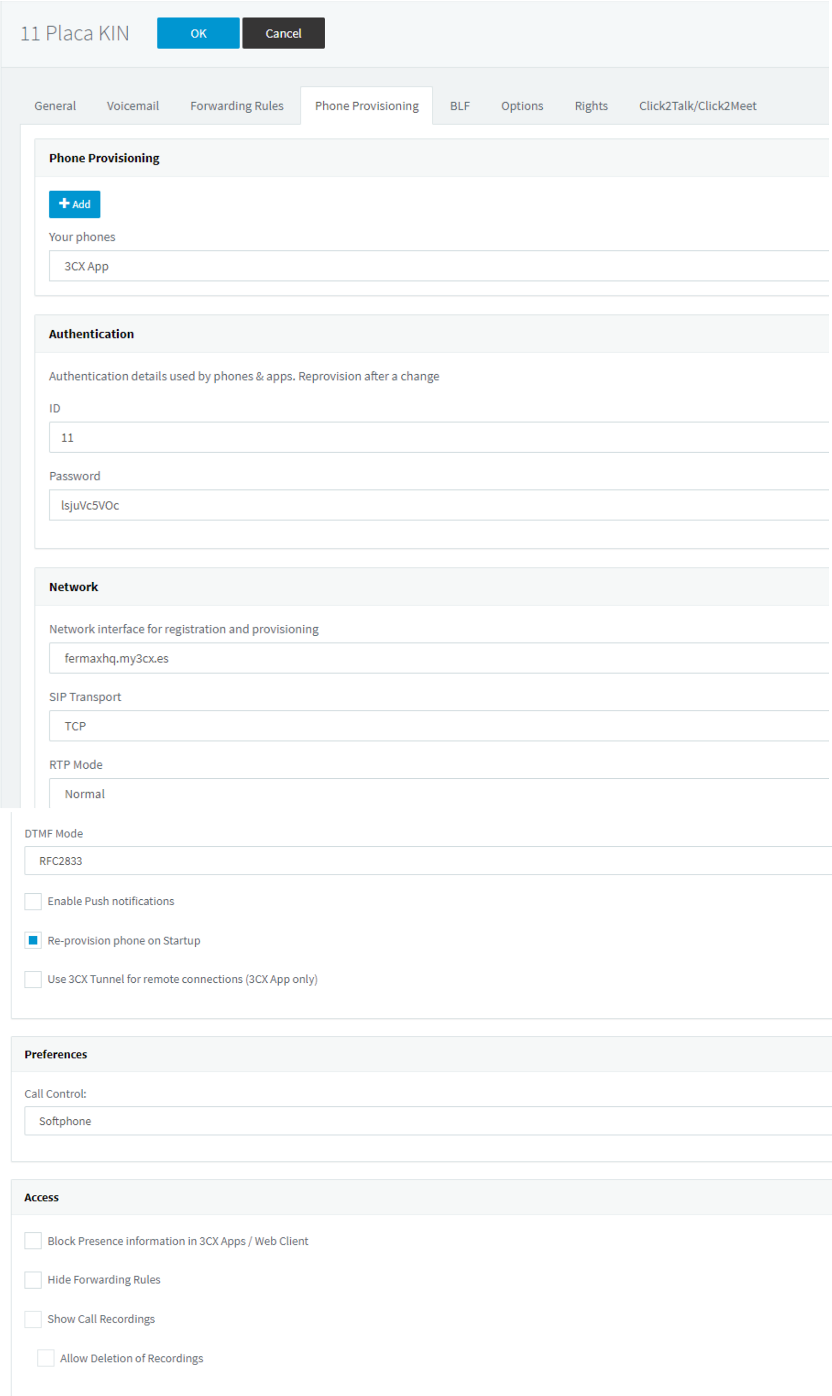
In the 'Users' section we select the 'ADD' option to create the new SIP extension. In the 'General' tab we fill in the data related to the street panel:



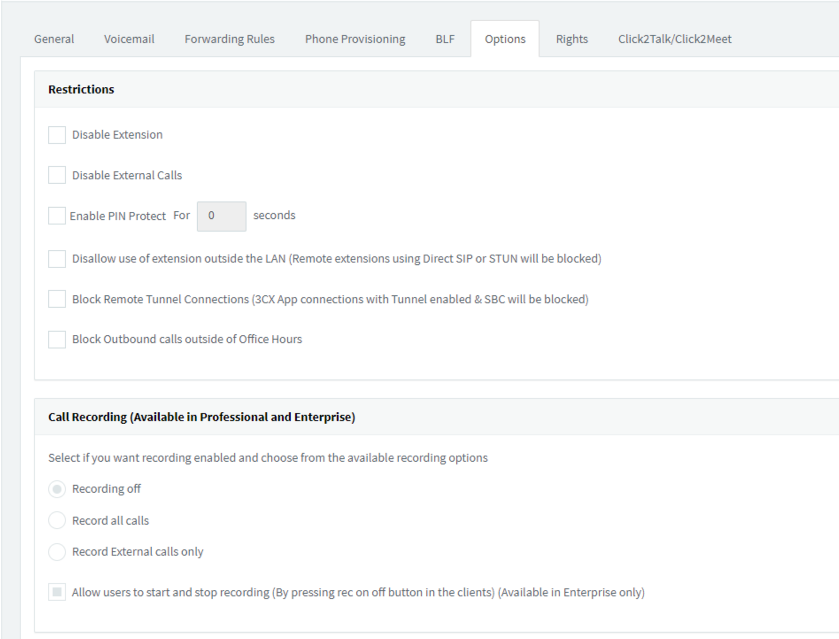
The extension number is filled in automatically. Uncheck the option ‘Enable Web Client’.

The accreditation details are provided in the 'Phone provisioning' tab:

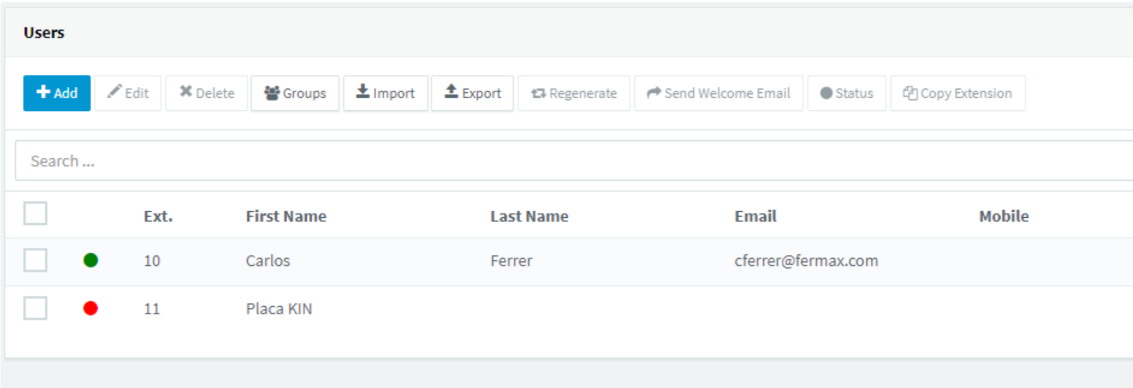
- ID. You can type the extension number, but a lack of security message will appear in the list of created extensions. This is the User to configure in the MEET board.
- Password. A password is generated automatically. It is recommended to keep it and use it in the configuration of the MEET board.
- Network interface. It is the URL of the SIP server to configure in the MEET board.
- Protocol: TCP.
- DTMS: RFC2833.
- Other details as shown below:



In the 'Options' tab you must uncheck the 'Block Remote Tunnel Connections' option:



Click on OK and the created extensions will be displayed:



The first (10) is the one that 3CX automatically creates, to test the web client, interesting to check that everything is configured correctly. The green ball indicates that this client has already registered

The second (11) is the one we just created for the MEET board. It is shown with a red ball because the panel has not yet been registered in the SIP PBX. It is the next step.

ENTRANCE PANEL CONFIGURATION

To configure the entrance panel, you must 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 username (admin) and default password (123456).

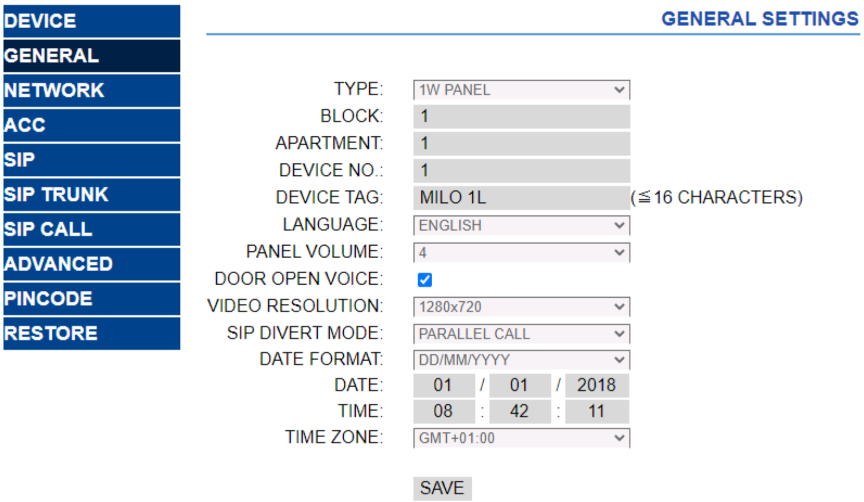
The following steps will be followed:

1) General settings

a. One way panel.

The panel must be configured as an Individual Panel, assign a block (by default 1), a dwelling (1, although by default it is 101), and a panel number (by default 1).

The device tag is important to identify the origin of the call in the SIP extensions. The resolution of the camera will be adjusted to the needs of the terminals used to receive the call.

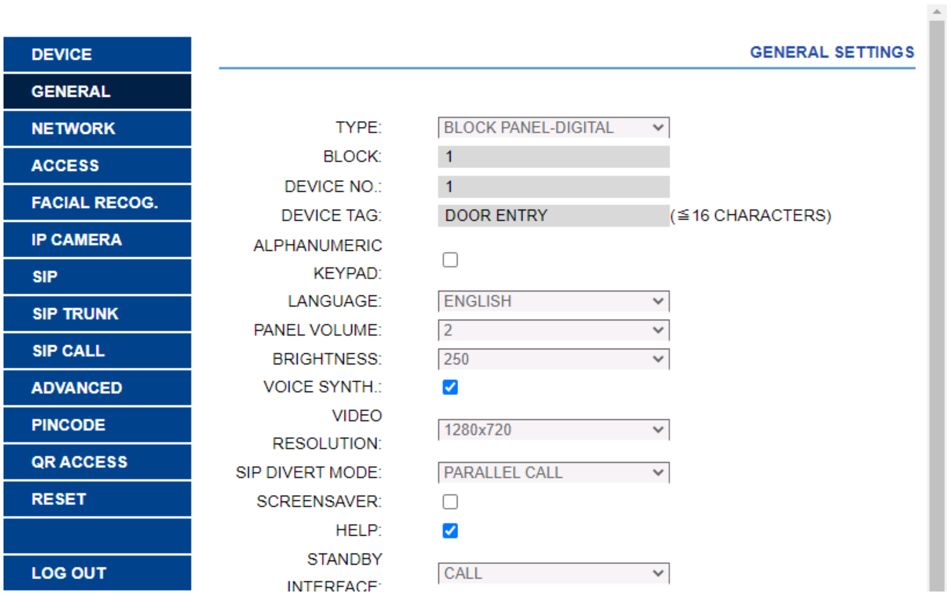


b. Building panel

In this example we have used a KIN entrance panel. The panel must be configured as a Block Panel, assign a block (by default 1) and a panel number (by default 1).

The Device Tag is important to identify the origin of the call in the SIP extensions.

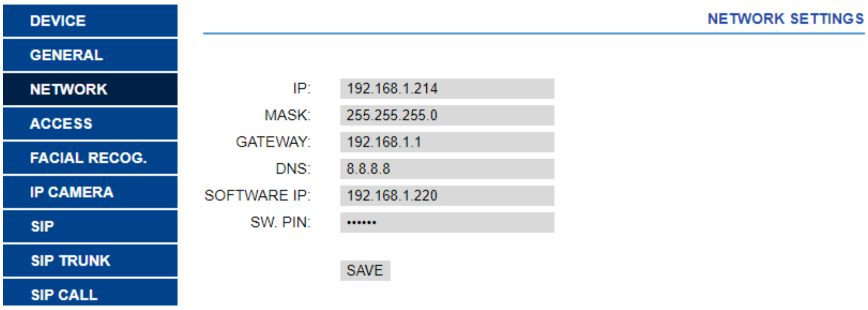
The resolution of the camera will be adjusted to the characteristics of the terminals used to receive the call.



2) Network configuration

An IP address compatible with the computer network of the installation will be assigned and the address of the Gateway or router to access the Internet will be indicated. A DNS server is necessary if a URL is to be used to access the PBX, this is our case.

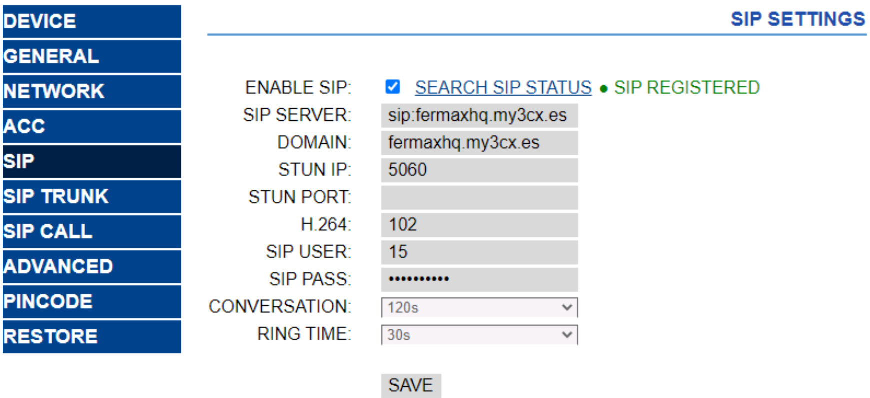
The software IP will be left by default because in this case it will not be used unless it is required to manage access credentials for employees (proximity cards or facial recognition). In this case, the IP address of the computer that has the MEET management software (MMS) installed will be indicated.



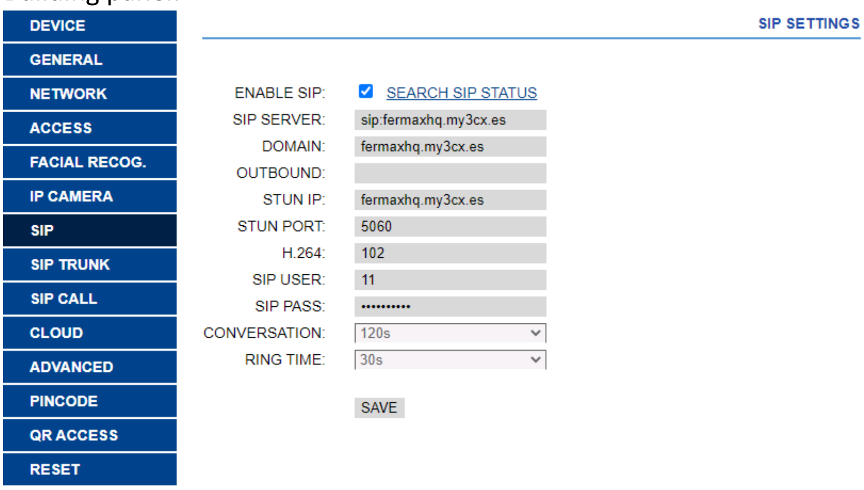
3) Switchboard configuration

In the SIP Configuration section, the URL of the assigned 3CX PBX will be indicated, as shown in the screen below. The username and password assigned to the entrance panel extension will be entered. Next, it will be checked if the panel is correctly configured in the control unit by clicking on the VIEW SIP STATUS link. If REGISTERED does not appear, check the information entered.

1 way panel:



Building panel:



In the console of the SIP PBX, the extension of the MEET board will already appear as registered:

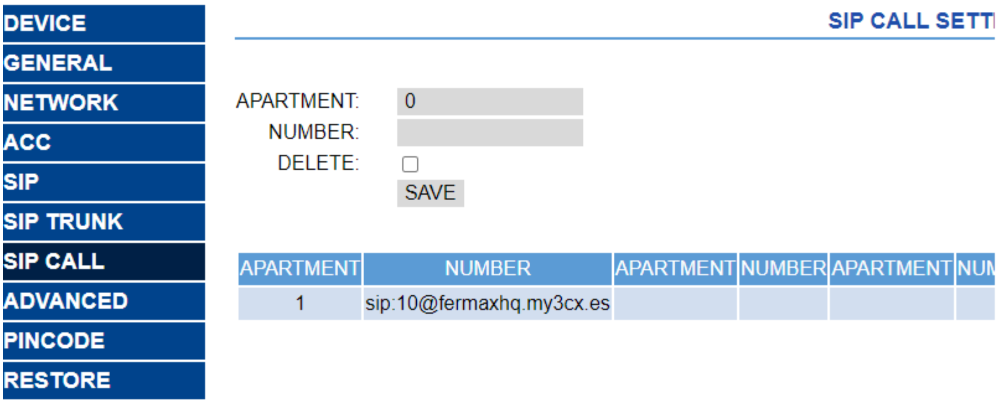
		Ext.	First Name	Last Name
		10	Carlos	Ferrer
		11	Placa	KIN

The exclamation mark warns that the assigned username (equal to the extension) does not offer an adequate level of security. If it is decided to change, it must be updated again at the street panel.

4) Call extensions configuration

a. One way panel.

You must go to the 'SIP CALL' section and enter the apartment number indicated in the 'GENERAL' section (1 in our example) and as NUMBER the SIP extension you want to call (10 in the example, with the format sip:extension@url_server, and click on 'SAVE'.



b. Building panel.

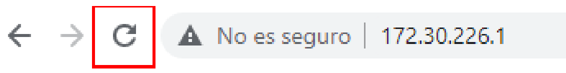
A CSV will be configured with the assignment of calling codes to the extensions to be called. The CSV file format can be downloaded from the SIP CALL section of the panel's web server, EXPORT option.

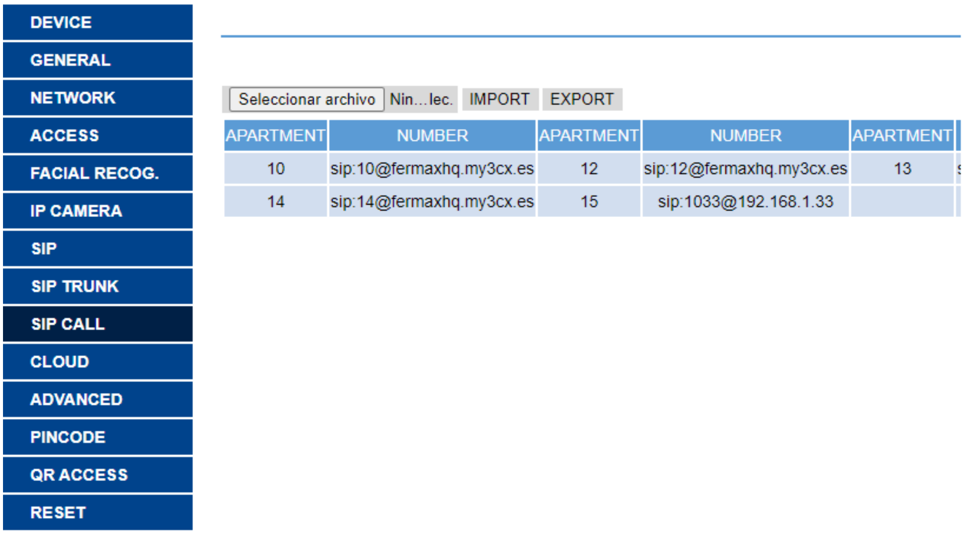
One example of CSV file is the following:

APARTMENT,NAME,		
10, sip:10@fermaxhq.my3cx.es		
12, sip:12@fermaxhq.my3cx.es		
13, sip:13@fermaxhq.my3cx.es		
14, sip:14@fermaxhq.my3cx.es		

First, indicate the code to be dialled and, separated by a comma, the extension of the switchboard that should receive the call when this call code is dialled. In the example, dialling 10 will call extension 10. It is possible to call more than one extension by listing multiple extensions in the same row separated by semicolon. One of these extensions can be a MeetMe license to be able to call a smartphone through the MeetMe application.

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



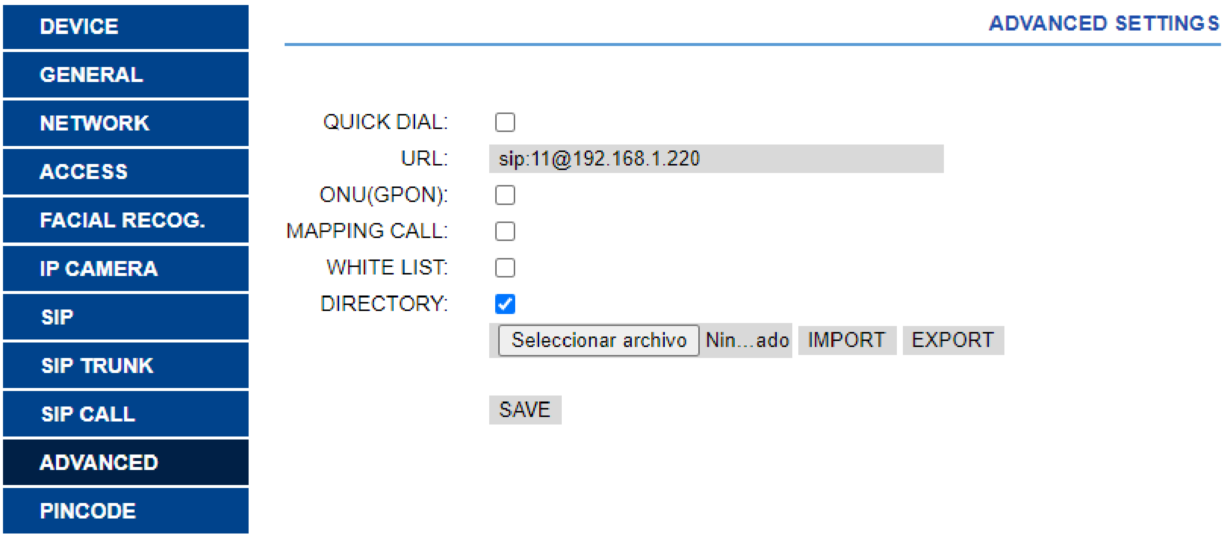


If we want to be able to call through the agenda in the case of a KIN panel, the agenda will be created in a CSV file (different one). The CSV file format can be downloaded from the ADVANCED section of the panel's web server, EXPORT option.

One row is filled per agenda item, indicating the calling code, Name to appear, 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,

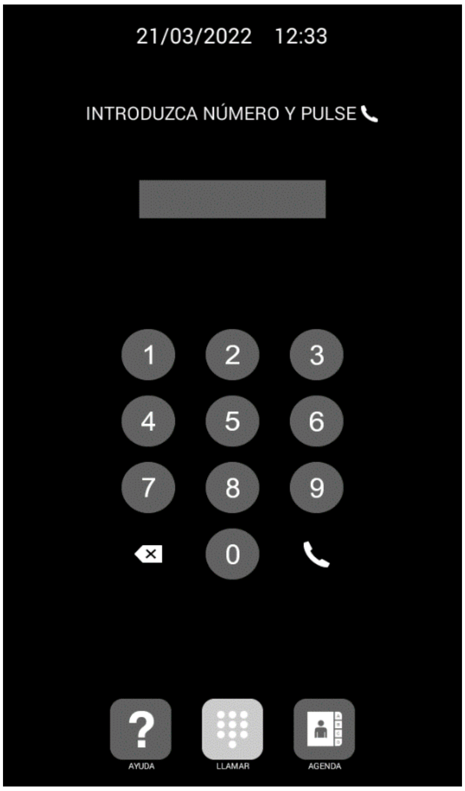
File is loaded using the IMPORT option:



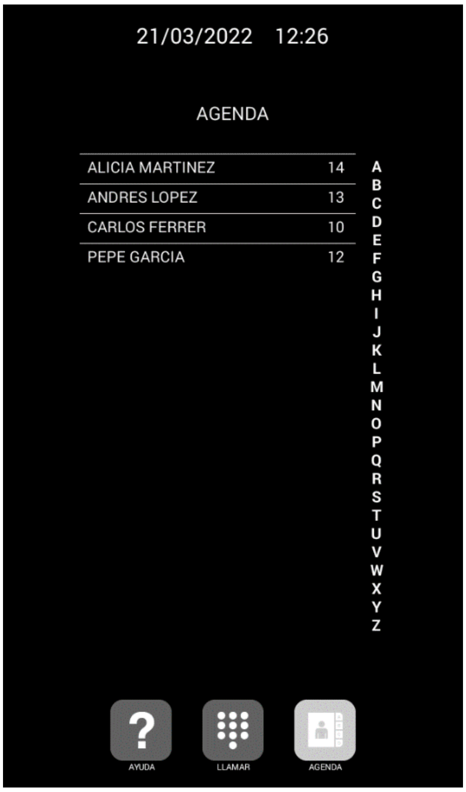
The AGENDA option is enabled so that this option appears on the panel.

WORKING MODE

To call a specific extension, you must press the call button (1-line panel) or enter the associated call code (building panel) and confirm with the bell button (Milo, Marine) or pick up icon (KIN). In the case of the KIN or MARINE panel, you also have the option of making the call through the Agenda, searching for the recipient's name and pressing on it.

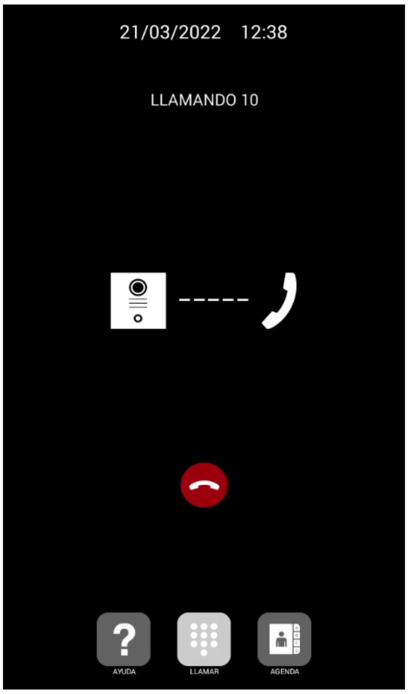


Direct call through code.



Call through Agenda.

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



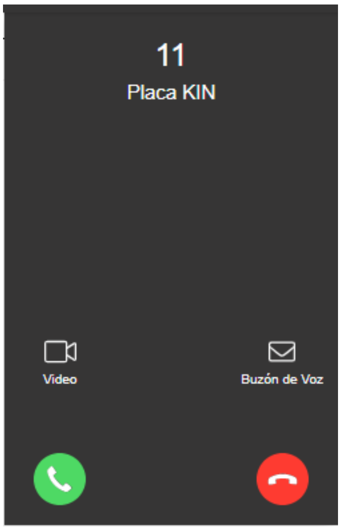
The called extension will receive the call tone and information on the origin of the call showing the Tag defined on the entrance panel and the image of the camera if it has the option of receiving video. The call will ring for a maximum of 30 seconds.

When accepting the call, bi-directional communication will be established that may last up to a maximum of 120 seconds.

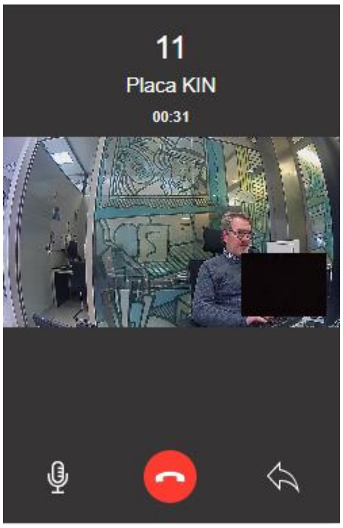
You have the option of unlocking the door by pressing the '*' or '#' key, or if it has been configured, the opening icon, which will simultaneously activate the first external relay of ref. 1490. The '0' key exclusively activates the second relay of ref. 1490.

Call reception examples:

- <u>3CX Web Client:</u>



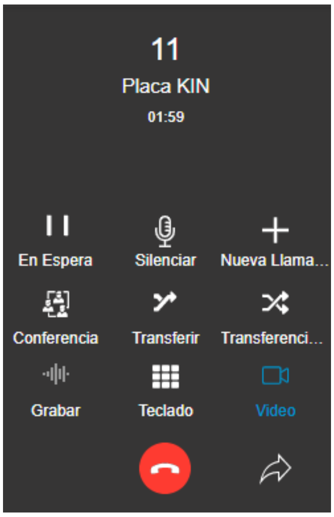
Call reception. It must be picked up first. It asks us if we want to receive video. Answer yes.



You get audio in both directions.

You can zoom to full screen of the video by clicking on the image.

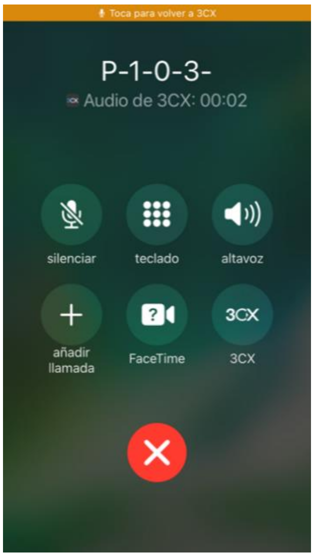
Other options are activated by pressing the left arrow.

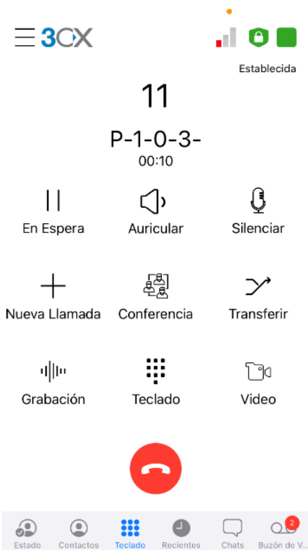


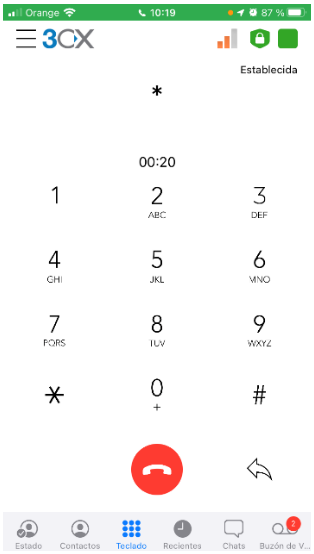
- Door opening: select keypad and press * or #.
- On hold: plays melody on panel, but then conversation cannot be regained.
- Mute: mute.
- Transfer: Transfer to another extension.
- You can record the conversation.

o 120" conversation time regulated by the street panel.

- <u>3CX App</u>
  - oThe call is received in App like Callkit and allows to open in 3CX app. The call is received whether the app is closed, in the background or open (in this case it does not go through callkit).







Callkit reception *App open* *Keypad activation for unlocking.*

  - 120" conversation time regulated by the street panel.
  - Limitations:
    - If the camera is activated, it does not show video because it is not compatible with the H264 video codec used by the MEET panel.
    - Hanging up the street panel does not end the conversation and continues with its timing.

- <u>Akuvox SIP telephone</u>

  o You can choose the reception with video or only audio. Choose video to view the visitor.

  o Two-way audio and video conversation is established.

- Door opening by pressing * or #.
- 120” conversation time regulated by the street panel.

OTHER CONSIDERATIONS

<u>MEET monitor compatibility:</u>

If you have a MEET monitor, you can receive the call simultaneously on the called extension and on the monitor. The monitor must have the call code dialled on the street panel assigned as the home number. The first to answer the call will cut off the reception on the other.

Another option is to configure the monitor as another extension of the switchboard, allowing it to be called from any extension (app, web client, SIP phones) or call those extensions from the monitor using the 'Extercom' option, entering the extension number. **In this mode, the video display is not shown until the call is answered by going off-hook and with some devices the video may be slow depending on the resolution.**

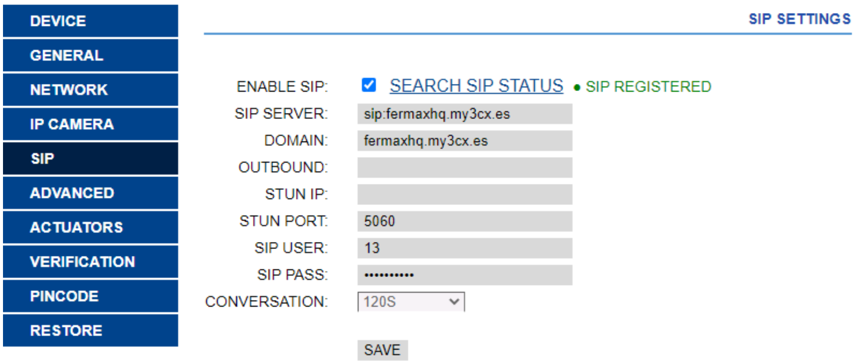
For this modality, a new extension must be registered in the SIP switchboard with the same configuration that was given to the street panel:

+ Add | Edit | Delete | Groups | Import | Export | Regenerate | Send Welcome Email | Status | Copy Extension

Search ...

		Ext.	First Name	Last Name	Email	Mobile
☐	●	10	Carlos	Ferrer	cferrer@fermax.com	
☐	●	11	⚠ Placa	KIN		
☐	●	12	Akuvox	IPphone		
☐	●	13	⚠ Monitor	WIT		
☐	●	14	Linphone	App		

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



Auto-on:

The connection from a SIP terminal to the entrance panel is made through a call to its extension number (10 in our example).

In SIP telephones it is necessary to disable the call in anonymous mode: