

IPefono

User Manual



ConectaIP

www.conectaip.es

Copyright ©

All rights reserved.

It is strictly forbidden to copy, reproduce or translate all or part of this document without the prior written consent of *ConectaIP*.

Changes

Improvements and/or changes may be made to the products and/or programs described in this publication at any time and without prior notice.

Trademarks

The brands and names of products mentioned in this manual may be trademarks, registered trademarks or the intellectual property of their respective owners.

TABLE OF CONTENTS

1. Introduction	1
1.1 Models	1
1.2 Manual guide	3

2. Installation	5
2.1 Dimensions	5
2.2 Power supply	5
2.3 Connections.....	6
2.4 Requirements	8
2.5 Startup	9
2.5.1 Assigning an IP address.....	10
2.6 Viewing the pin assignment.....	12

3. Operation	13
3.1 Operating mode	13
3.2 Modes	13
3.3 Keyboard	14
3.4 Playing messages	14
3.5 Inputs and outputs	14
3.6 Audio sensor.....	15
3.7 Video.....	15

4. Setup Wizard.....	17
4.1 Initial configuration.....	17
4.2 General settings	18
4.3 Network configuration.....	18
4.4 Connection to a SIP server.....	18
4.5 Direct call to a VoIP telephone	19
4.6 Acoustic echo cancellation	19
4.7 Audio volumes	19
4.8 Open a door or barrier by keying in a code	20
4.9 Paging functionality	20
4.10 Video camera module.....	20
4.11 Paging of the auxiliary audio input	21

5. Configuration	23
5.1 Initial configuration.....	23
5.2 General settings	25
5.3 Network configuration.....	26
5.4 Setting the time.....	27
5.5 Input and output ports.....	28
5.5.1 Audio sensor.....	30

5.6	Voice over IP settings	31
5.6.1	SIP protocol. VoIP connections	33
5.6.1.1	Point-to-point	34
5.6.1.2	PBX	35
5.6.1.3	Internet VoIP services	36
5.6.1.4	Gateway to the telephone network	37
5.6.2	Acoustic echo canceller	38
5.6.2.1	Adaptive echo cancellation	38
5.6.2.2	Echo cancellation through attenuation	39
5.6.2.3	Half-duplex echo cancellation	39
5.6.2.4	No echo cancellation	40
5.6.3	Background noise canceller	41
5.6.4	Paging	41
5.6.5	Messages	42
5.7	Video settings	43
5.8	Keyboard settings	45
5.8.1	Making calls	46
5.8.2	Activating outputs	46
5.8.3	Playing messages	46
5.8.4	Activating emergency mode	47
5.9	Emergency Mode Settings	48
5.10	Auto-configuration	50

6- System administration 51

6.1	Entering system administration	51
6.2	Administration options	52
6.3	Configuration file management.....	53
6.3.1	Saving the configuration	53
6.3.2	Configuration list.....	53
6.3.3	Editing the configuration file	57
6.4	Updating firmware	58
6.5	Estado del dispositivo.....	60
6.6	Viewing the video	61
6.7	Customising messages	62
6.8	Administration and diagnostics tools	63
6.8.1	Traces.....	63
6.8.2	Network commands.....	65
6.8.3	Audio commands.....	65
6.8.4	Video commands.....	65

7. Drawings and dimension	67
7.1 IPefono Wall Mount	67
7.2 IPefono Wall Mount Embedded.....	69
7.3 IPefono HQ with enclosure.....	70
7.4 IPefono LC with enclosure.....	71
7.5 IPefono HQ and LC OEM	72
7.6 Camera	73

1

Introduction

IPefono is a range of intercom devices that enable voice communication over IP networks.

1.1 Models

They can be divided into two categories:

Models designed for end users: IPefono Wall Mount (WM) for surface mounting, IPefono Wall Mount with Keyboard (WMT) and IPefono Wall Mount Embedded (WMe) for embedded mounting.



FIGURE 1. IPEFONO WM, WMT AND WME

Models for integrators: IPefono LC and IPefono HQ, either in the OEM version (electronic board only) or with its respective enclosure. In these cases the speakers and microphone have to be connected externally.



FIGURE 2. IPEFONO LC AND HQ

The IPefono LC / WM / WMT /WMe has:

- ❑ A microphone input.
 - ❑ A 4Ω or 8Ω speaker output. A class-D amplifier with 3.2 W.
 - ❑ A call progress indicator.
 - ❑ 5V power supply.

The communication ports are as follows:

- ❑ 100BT Ethernet port with remote power supply (Power over Ethernet).

It also has the following I/O ports (inputs and outputs):

- ❑ 1 input (call button) with wire break detector.
- ❑ 1 output (opening of doors).
- ❑ Connector for 4x4 matrix keyboard.

The IPefono HQ has:

- ❑ A microphone input.
- ❑ A 4Ω or 8Ω speaker output. A class-D amplifier with 3.2 W.
- ❑ A call progress indicator.
- ❑ 5V to 24V auxiliary power supply.

The communication ports are as follows:

- ❑ Two Ethernet ports (Switch Ethernet).

It also has the following I/O ports (inputs and outputs):

- ❑ 1 input (call button).
- ❑ 2 inputs or 2 outputs (opening of doors).
- ❑ Connector for 4x4 matrix keyboard.

1.2 Manual guide

This manual explains every aspect of the startup, operation, configuration and administration of the IPefono. It is divided into the following chapters:

- ❑ Chapter 1. Introduction.
- ❑ Chapter 2. Installation and startup of the device.
- ❑ Chapter 3. Operating Mode.
- ❑ Chapter 4. Configuration Assistant simplifies the task of configuring, deploying, and administering the device.
- ❑ Chapter 5. Configuration with the web browser, enabling:
 - The configuration of the system's general settings.
 - The configuration of the network settings.
 - The setting of the time.
 - The configuration of the settings of the input and output ports.
 - The configuration of the Voice over IP settings.
 - The configuration of the video.
 - The configuration of the keyboard.
 - The configuration of the emergency mode.
- ❑ Chapter 6. System administration, which basically explains how to manage the configuration files and update the firmware.

2

Installation

2.1 Dimensions

The physical dimensions of the different models of IPefono are as follows:

IPefono WM /WMT 172 x 103 x 32 mm

IPefono LC 85 x 55 x 25 mm

IPefono HQ 70 x 90 x 57 mm

2.2 Power supply

The IPefono HQ can be powered externally from 5V to 24V (using a transformer that is not included with the product).

In the case of the IPefono LC / WM / WMT /WMe with PoE it can receive power over the Ethernet or from a 5V local power supply (using a transformer that is not included with the product).

2.3 Connections

The IPefono LC has the following different connections:

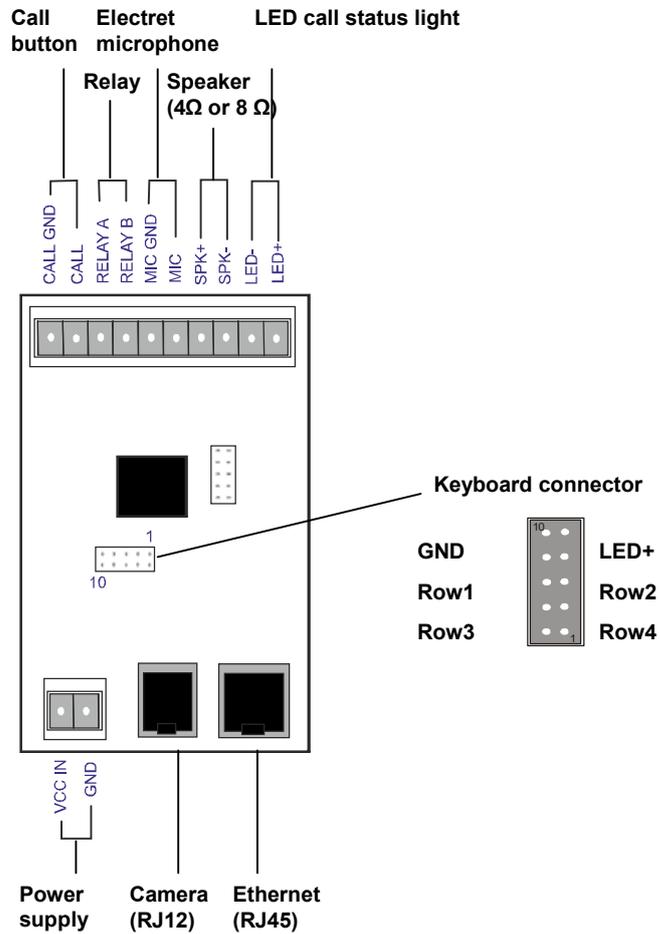


FIGURE 3. IPEFONO LC CONNECTIONS

The IPefono HQ has the following connections:

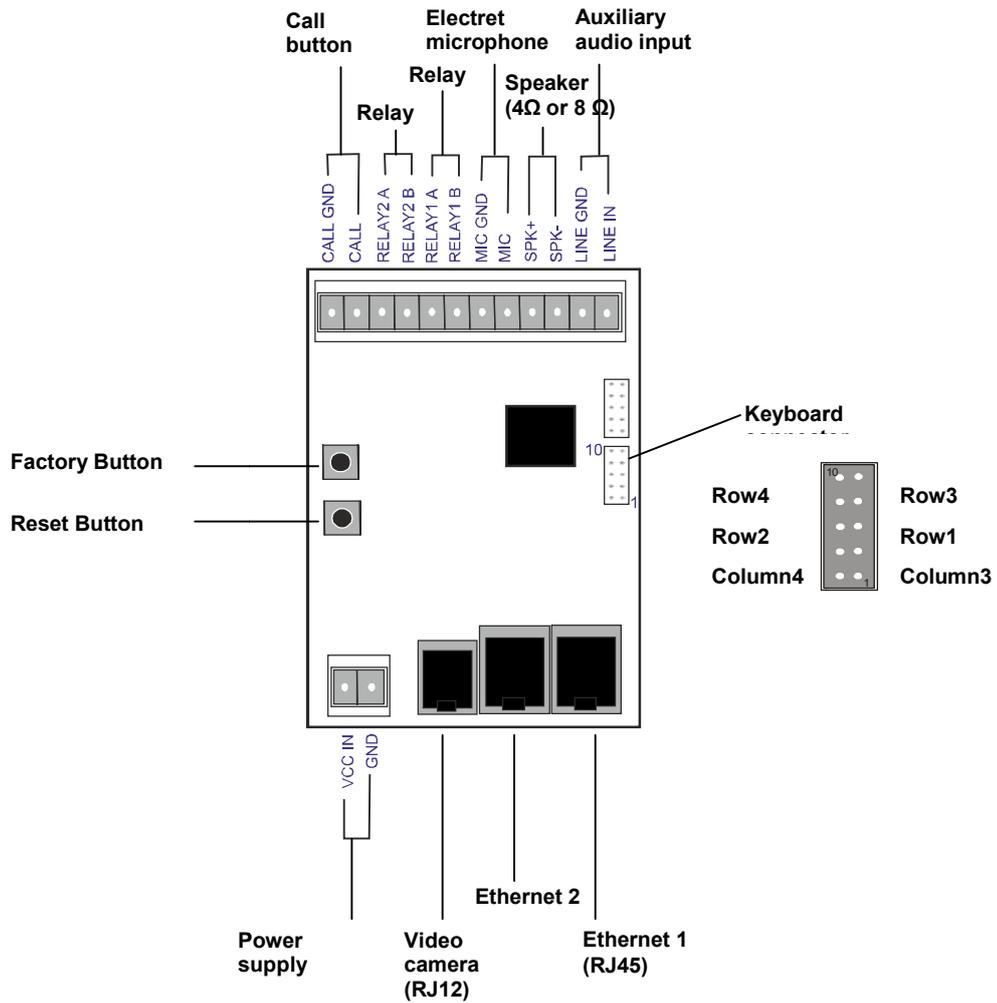


FIGURE 4. IPEFONO HQ CONNECTIONS

2.4 Requirements

To configure an IPefono you need:

- ❑ A PC with any operating system that has a web browser, with a connection to a local network by Ethernet or Wi-Fi.
- ❑ An IPefono.
- ❑ Optionally a switch, depending on the scenario chosen. This must be Power over Ethernet if the IPefono PoE has no local power supply.
- ❑ One or two Ethernet cables, depending on the scenario chosen (see [Figure 5](#)) with RJ45 connectors.

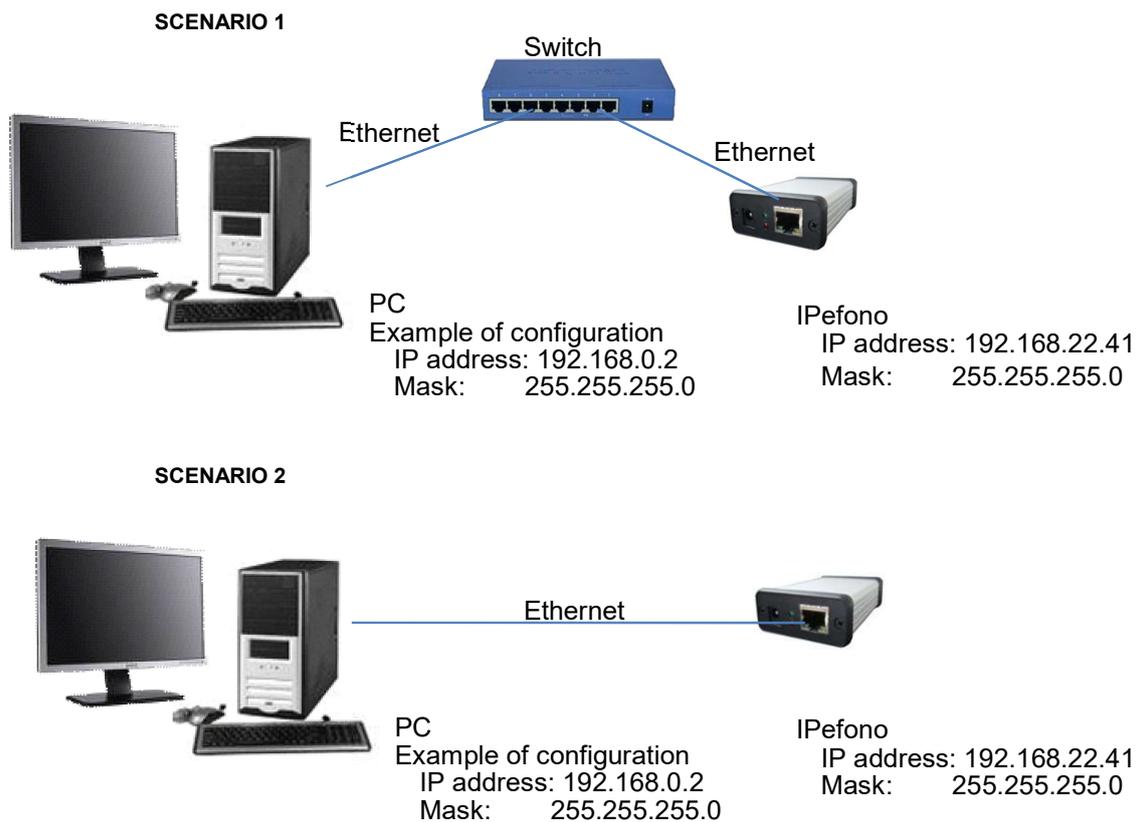


FIGURE 5. CONFIGURATION SCENARIOS

2.5 Startup

The IPefono leaves the factory with a pre-set IP address, either **192.168.22.41** or **192.168.1.61** (mask 255.255.255.0), so that it can be connected to the IP network. It can be configured in two different ways:

- ❑ With the PC and IPefono connected to the local network (via a switch).
- ❑ With the PC and IPefono directly connected (via the Ethernet connection).

When the connection has been established according to one of the above scenarios (see [Figure 5](#)) enter the device's IP address in the address bar of your browser, if it has not been changed:

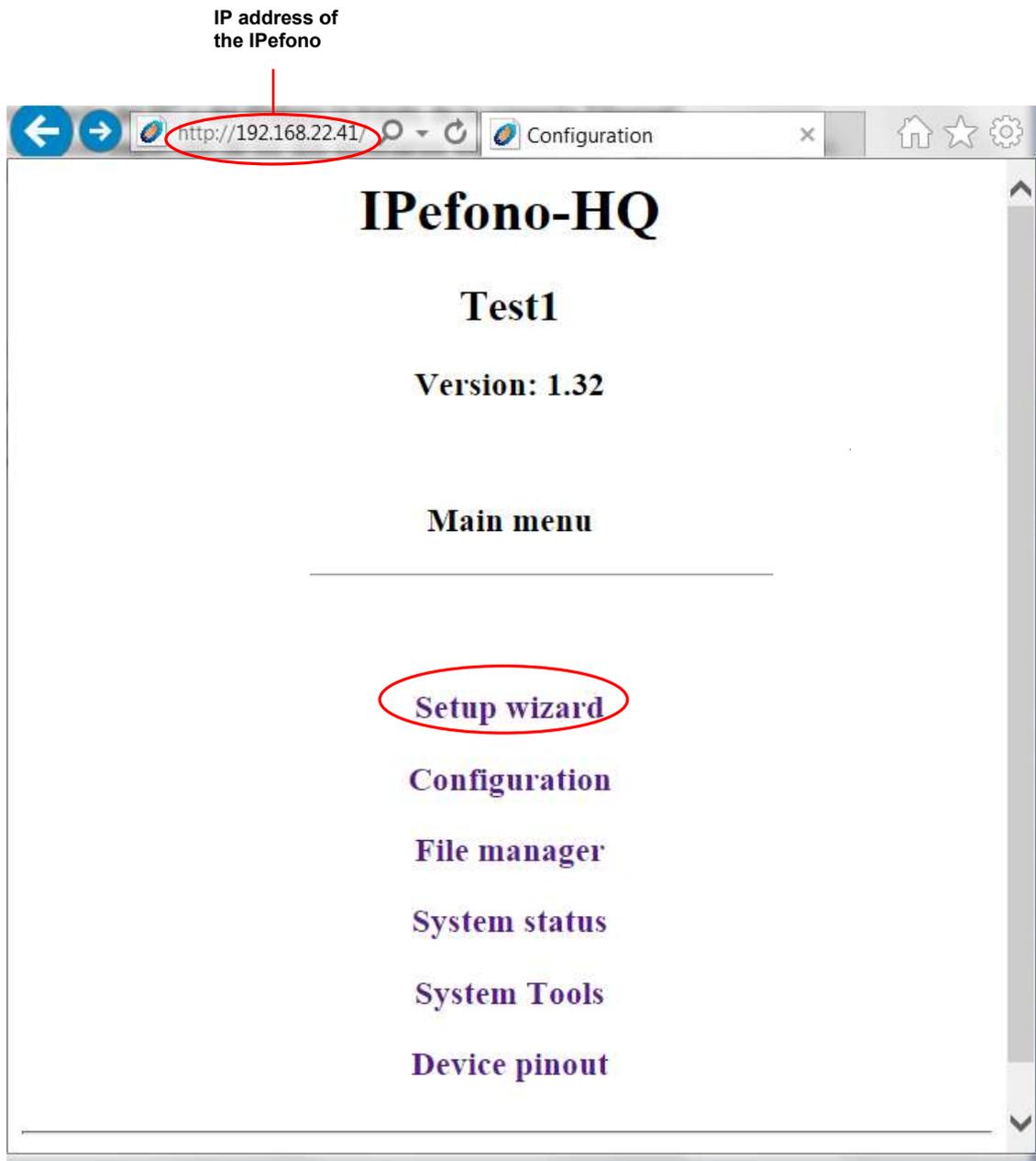


FIGURE 6. ACCESSING SETUP WIZARD WITH THE WEB BROWSER



The PC needs to have access to the device's IP network. Change the PC's IP address where necessary.

2.5.1 Assigning an IP address

Firstly, the device must be assigned a valid IP address from your network. Navigate to the “**Setup wizard**” option shown in [Figure 6](#) and then in the new window ([Figure 7](#)) “**Network parameters**”.

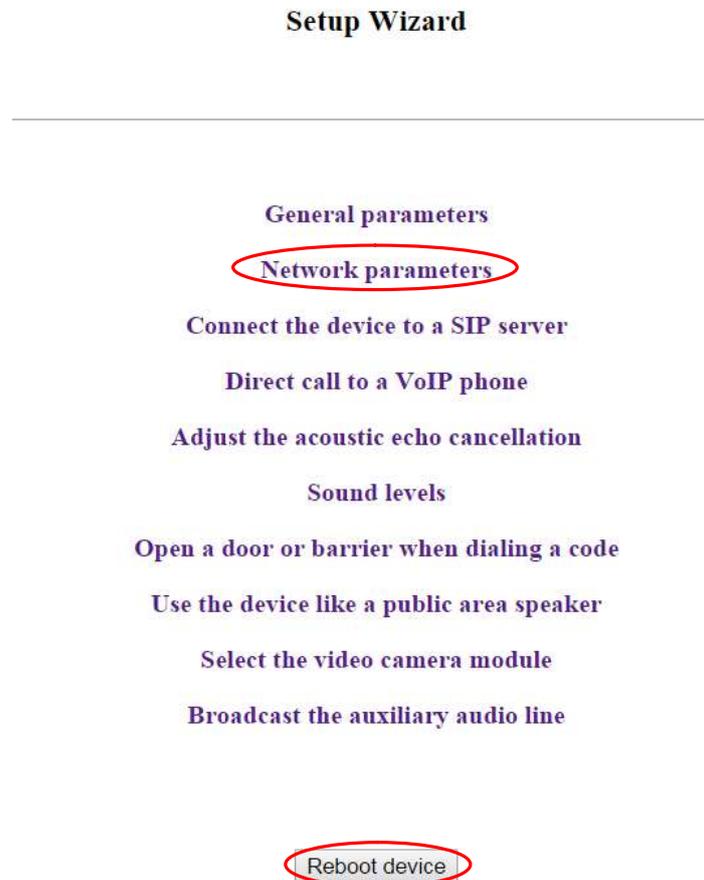


FIGURE 7. ACCESS TO THE DEVICE’S IP ADDRESS

Configure the new IP address:

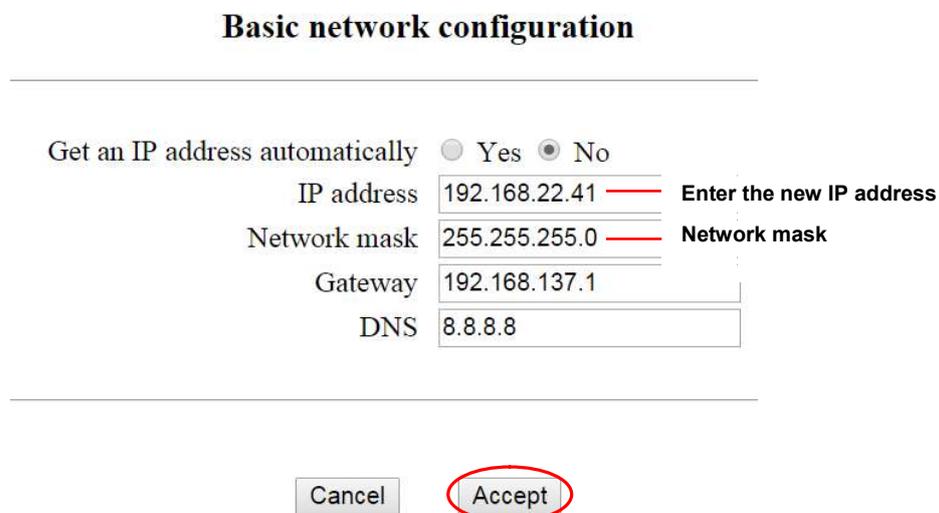


FIGURE 8. CONFIGURATION OF THE IP ADDRESS

Finally, you have to reboot the device for the new IP address to come into effect. Select the option “**Reboot device**” shown in [Figure 7](#).



If you press the IPefono HQ’s factory button (see Figure 4 for its location), the device says its IP address.

2.6 Viewing the pin assignment

Select the “*Device pinout*” option on the home screen (see [Figure 6](#)) to view the IPefono’s pin assignment.

Audio line input	IPefono HQ	Ethernet 1
Audio line GND		Ethernet 2
Speaker -		Video camera
Speaker +		
Electret microphone input		System led
Electret microphone GND		Power supply GND
Relay 1 contact A		Power supply 5V to 24V
Relay 1 contact B		
Relay 2 contact A		
Relay 2 contact B		
Call button contact A		
Call button contact B		

FIGURE 9. VIEWING THE PIN ASSIGNMENT

3

Operation

3.1 Operating mode

The IPefono allows you to establish a voice communication over IP networks.

When you press the user's call button the device calls the IP address configured in "**Destination SIP URL**" (see [Figure 35](#)). If this is not in operation it calls the IP address configured in "**Alternative SIP URL**". If this fails, it gives the message configured in "**No response from the Helpdesk**" (see [Figure 48](#)).

There are various different operating modes:

- ❑ **Press to speak.** Hold down the call button to speak to the destination.
- ❑ **In open conversation.** You talk without having to hold down the call button.
- ❑ **Paging system.** You simultaneously talk to several devices that make up a paging zone, selected using a group code.
- ❑ **Broadcast.** A call to one of the intercoms can be played simultaneously on all intercoms from the same group.
- ❑ **Listen.** You can call an intercom to listen to what is happening in the zone.
- ❑ **Play messages.**
- ❑ **Emergency mode.** When the intercom is in this mode it automatically plays a pre-set message and tones which, in turn, can be resent to the other devices in the same paging zone.

This system also allows you to remotely activate outputs (for example, to open a door) and the reading of inputs.

3.2 Modes

The device can operate with:

- ❑ An echo suppressor. Cuts off the return audio when the speaker emits a signal to prevent the sound emitted from reaching the speaker.
- ❑ Background noise suppressor. Cuts off the audio when noise is detected to prevent background noise from being transmitted.

3.3 Keyboard

If you have an IPefono with a keyboard, this offers the following functionalities:

- ❑ Make calls (see section 5.8.1).
- ❑ Activate outputs (section 5.8.2)
- ❑ Play messages (section 5.8.3)
- ❑ Activate emergency mode (section 5.8.4)

3.4 Playing messages

The system allows you to play messages (see 6.7 Customising messages). These messages can also be played by connecting to the device and sending the command from the web browser. To do this, key the IP address of the device in the address bar of your browser, followed by:

http://(IP address)/play?(name of the message file to be played)

For example <http://192.168.22.41/play?OutofService.wav>

3.5 Inputs and outputs

You do not need to activate anything to be able to use these signals, simply use them on the configuration screen for the desired function, as described in each section. However, there are a number of direct actions.

The inputs can:

- ❑ Activate an output when on ("**Activate this output when ON**", see Figure 30).
- ❑ Play a pre-recorded message ("**Play a message when this input is ON**", see Figure 30).
- ❑ Lock an output ("**Lock output meanwhile this input is ON**", see Figure 31), i.e. not allow the output to be deactivated until the input has been deactivated (e.g. to prevent a barrier from coming down while somebody is passing underneath).

The outputs are controlled as follows:

- ❑ They can be automatically deactivated when a configurable time has elapsed ("**Automatic deactivation time in milliseconds**", see Figure 31).
- ❑ They are activated when the user enters a pre-set code on the keyboard or on the remote device that is communicating with it ("**Activate when the user type this code**", see Figure 31).
- ❑ They are also deactivated when the user enters a pre-set code on the keyboard or on the remote device that is communicating with it ("**Deactivate when the user type this code**" see Figure 31).
- ❑ Using an HTTP command (<http://192.168.22.41/SetOutput?20000>) see Table 1. .

3.6 Audio sensor

The audio sensor is activated when it detects a certain level of noise during a specified minimum amount of time. It is configured using the “**audiosensor**” option shown in [Figure 29](#).

IPefono’s sensor is designed to detect noises and voices that exceed a specific threshold level and duration. So, for example, you can connect the intercom system to a central alarm system so that it can detect noises in the area or you can make the intercom call when someone approaches it and speaks.

The level is configured in the “**Audio sensitivity threshold**” setting (see [Figure 32](#)) and you can select 5 levels of sensitivity. To prevent the sensor from constantly being triggered, you should enter a minimum amount of time during which the sensor has to detect noise. This is done using the “**Time threshold in milliseconds**” setting (see [Figure 32](#)).

When the sensor is triggered, you can perform the following actions:

- ❑ Launch a call to the destination, pre-configured in the VoIP/SIP environment (see [Figure 35](#)).
- ❑ Play a message that has been pre-recorded on a “wav” file (see [Figure 32](#)).
- ❑ Activate one of the outputs (see [Figure 32](#)).

3.7 Video

It is possible to connect two models of cameras to the intercoms. They basically have different lenses and sizes. On the one hand, there is the C429 model, which can be screwed onto the front and you can select the most suitable lens.



FIGURE 10. C429 MODEL

The other camera model, C339, is much smaller and you need to use hot-melt adhesive or glue to attach it to the front as there are no holes for this. It has a pinhole lens which allows it to be discreetly placed on the front. This camera has an operating mode (C339-SPI) that allows you to capture up to 10 images per second.



FIGURE 11. C339 MODEL

For the WM / WMT / WMe models, the C339 model is always used.

See section [5.7 Video settings](#)

4

Setup Wizard

4.1 Initial configuration

The Setup Wizard simplifies configuring the IPefono, therefore allowing you to configure the device's basic settings without having to enter the web configurator which contains all of the configuration settings.

Navigate to the device's setup wizard by following the steps that are outlined in section [2.5 Startup](#). There are the following different hyperlinks on the setup wizard home screen

General parameters	<i>General settings</i>
Network parameters	<i>Network configuration</i>
Connect the device to a SIP server	<i>Connect the device to a SIP server</i>
Direct call to a VoIP phone	<i>Make a direct call to a VoIP telephone</i>
Adjust the acoustic echo cancellation	<i>Adjust the echo cancellation settings</i>
Sound levels	<i>Audio volumes</i>
Open a door or barrier when dialing a code	<i>Open a door or barrier by inputting a code</i>
Use the device like a public area speaker	<i>Use the device's paging feature</i>
Select the video camera module	<i>Select the video camera module</i>
Broadcast the auxiliary audio line	<i>Retransmission of the auxiliary audio line</i>

FIGURE 12 . SETUP WIZARD HOME SCREEN

4.2 General settings

If you go to the setup wizard home screen (Figure 7), by clicking on the “**General parameters**” option, you can configure the following settings:

General parameters

Name of this device	<i>Name of the device</i>
Description to display in the remote phone	<i>Description of the device displayed on the remote telephone</i>
Device administration username	<i>Username</i>
Device administration password	<i>Password</i>

FIGURE 13. CONFIGURATION OF GENERAL SETTINGS

The initial configuration system has unrestricted access, if you want authentication to be required to enter, complete the username and password fields.

4.3 Network configuration

The network is configured by first selecting the “**Network parameters**” option (see Figure 7). The settings are as follows:

Network parameters

Get an IP address automatically	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Whether the parameters are assigned from a DHCP server</i>
IP address		<i>IP address of the device</i>
Network mask		<i>Mask of the IPefono</i>
Gateway		<i>IP address of the gateway</i>
DNS		<i>DNS port</i>

FIGURE 14. CONFIGURATION OF NETWORK SETTINGS

4.4 Connection to a SIP server

The connection to a SIP server is configured by first selecting the “**Connect the device to a SIP server**” option (see Figure 7). The settings are as follows:

Connect the device to a SIP server

SIP server hostname or IP address	<i>URL or IP address of the server</i>
Account number	<i>Account number set up on the server</i>
Account password	<i>Password to log into the server</i>
Destination call number	<i>Extension number that is called when clicking on the call button</i>

FIGURE 15. CONFIGURATION OF CONNECTION TO A SIP SERVER

4.5 Direct call to a VoIP telephone

If the IPefono is directly calling a VoIP telephone, go to the wizard home screen (Figure 7) and click on “**Direct call to a VoIP phone**” option. The settings are as follows:

Direct call to a VoIP phone

Phone extensión number	<i>Extension number allocated to the telephone</i>
VoIP phone IP address	<i>IP address of the telephone</i>
SIP port configured in the phone	<i>The telephone’s SIP port</i>

FIGURE 16. CONFIGURATION OF DIRECT CALL TO A VOIP TELEPHONE

4.6 Acoustic echo cancellation

The acoustic echo cancellation is configured by first selecting the “**Adjust the acoustic echo cancellation**” option (see Figure 7). The settings are as follows:

Adjust the acoustic echo cancellation

Type of cancellation depending on the ambient noise	↓	<i>Choose the most suitable option, depending on the ambient noise: disabled (no echo cancellation), soft (little ambient noise, for example indoors), hard (high ambient noise), extreme (operating in half adaptive duplex mode)</i>
---	---	--

FIGURE 17. CONFIGURATION OF ACOUSTIC ECHO CANCELLATION

4.7 Audio volumes

The audio volumes are configured by first selecting the “**Sound levels**” (see Figure 7). The settings are as follows:

Audio volumes

On communication	<i>Conversation volumen control</i>
On incoming call ring signal	<i>Ring volumen control</i>
On tone signal	<i>Tone volumen control</i>
Playing pre-recorded message	<i>Volume of pre-recorded messages</i>
Playing the signal from the auxiliary input	<i>Volume of the auxiliary line</i>

FIGURE 18. CONFIGURATION OF AUDIO VOLUMES

4.8 Open a door or barrier by keying in a code

If the IPefono opens a door or barrier by keying in a code, go to the wizard home screen (Figure 7) and click the “**Open a door or barrier when dialing a code**” option. The settings are as follows:

Open a door or barrier when dialing a code

Relay connected to the door	<i>Output number connected to the door</i>
Opening code in the remote phone	<i>Code keyed into the remote telephone to open it</i>
Opening code in the local keypad	<i>Code keyed in on the IPefono keyboard to open it</i>
Time during which the relay remain activated	<i>Time in seconds for which the output remains activated. '0' if it always remains activated</i>
Message to play when the door is opened	<i>Message played on activating the output</i>

FIGURE 19. CONFIGURATION OF OPEN A DOOR O BARRIER BY KEYING IN A CODE

4.9 Paging functionality

If you have an IPefono Speaker, its functionality is configured by first selecting the option “**Use the device like a public area speaker**” (see Figure 7). The settings are as follows:

Use the device like a public area speaker

Playing message when the broadcaster establish connection	<i>Select the message that is played when the paging connection is established</i>
Broadcaster connection timeout	<i>Maximum paging connection time. '0' there is no limit.</i>
Broadcast paging zone of this speaker	<i>Paging zone to which the IPefono belongs</i>

FIGURE 20. CONFIGURATION OF PAGING FUNCTIONALITY

4.10 Video camera module

The video camera module is configured by first selecting the “**Select the video camera module**” option (see Figure 7). The settings are as follows:

Select the video camera module

Camera module	<i>Model of the camera, see section 3.7 Video</i>
---------------	---

FIGURE 21. CONFIGURATION OF VIDEO CAMERA MODULE

4.11 Paging of the auxiliary audio input

The audio signal that is received through the auxiliary line can be delivered to a group by paging. The selection of the paging group is configured by first selecting the option “**Broadcast the auxiliary audio line**” option (see [Figure 7](#)). The settings are as follows:

Broadcast the auxiliary audio line

Broadcast paging zone

Selection of the paging group where the auxiliary audio input signal is played

FIGURE 22. CONFIGURATION OF PAGING OF THE AUXILIARY AUDIO INPUT

5

Configuration

5.1 Initial configuration

IPefono's configuration is based on an '.ini' text file, which can be edited locally with any text editor or while directly connected to the device via the web browser. No specific configuration software is ever needed.

This chapter covers all aspects of the configuration of the IPefono, using the web browser developed for that purpose. This user interface makes configuration easy as it contains intuitive menus that enable simple and flexible navigation in the system environment.

Navigate to the device's configuration screen by following the steps that are outlined in section [2.5 Startup](#). Select the option "**Configuration**" from the home screen (see [Figure 6](#)).

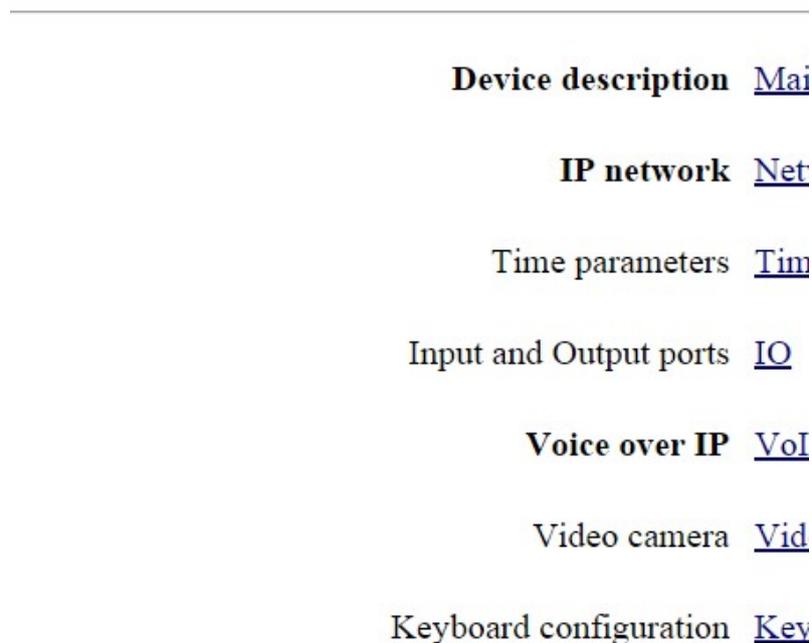


FIGURE 23. CONFIGURATION HOME SCREEN

There are the following different hyperlinks on the configuration home screen

IPefono

Device description	Main	<i>General configuration settings</i>
IP network	Network	<i>Network configuration</i>
Time parameters	Time	<i>Time settings</i>
Input and Output ports	IO	<i>Input and Output ports</i>
Voice over IP	VoIP	<i>Voice over IP settings</i>
Video camera	Video	<i>Video settings</i>
Keyboard configuration	Keyboard	<i>Keyboard configuration</i>
Emergency mode	Emergency	<i>Emergency Mode Settings</i>
Customize apps	Custom	<i>Special applications</i>

FIGURE 24. CONFIGURATION HOME SCREEN

5.2 General settings

If you go to the configuration home screen (Figure 23), by clicking on the “**Main**” option, you can configure the following system settings:

Main		
Device type	IPefono-HQ	
Firmware version	1.15	
Device name	4041	<i>Name of the device</i>
Device description	Input1	<i>Description of the device</i>
Administrator contact	_____	<i>IP address of the administrator</i>
Device location	_____	<i>Location of the device</i>
Administrator username	_____	<i>Username</i>
Administrator password	_____	<i>Password</i>
Auto set up is activated	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Auto-configuration activated o deactivated (see section 5.10 Auto-configuration)</i>

FIGURE 25. CONFIGURATION OF GENERAL SETTINGS

The initial configuration system has unrestricted access, if you want authentication to be required to enter, complete the username and password fields.

5.3 Network configuration

The network is configured by first selecting the “**Network**” option (see [Figure 23](#)). The settings are as follows:

Network		
Get network configuration from DHCP server	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Whether the parameters are assigned from a DHCP server</i>
Local network addresses	address(0) address(1)	<i>IP addresses of the device</i>
Router IP address	0.0.0.0	<i>Address of the router</i>
Domain name server	0.0.0.0	<i>Name resolution IP address</i>
Domain name	ConectaIP	<i>Domain name</i>
Telnet server port	23	<i>Telnet server port</i>
Web server port	80	<i>Web server port</i>
Domain Name Server port	53	<i>DNS port</i>
Network Time Protocol port	123	<i>NTP port (time synchronisation)</i>
Simple Network Management Protocol port	161	<i>SNMP port</i>
SNMP trap URL destination	_____	<i>SNMP trace sender address</i>
SNMP trap port destination	162	<i>SNMP trace destination port</i>
Virtual Network identifier (2-4095,0=disabled)	0	<i>VPN identifier</i>
Virtual Network priority (0-7)	0	<i>VPN priority</i>

FIGURE 26. CONFIGURATION OF NETWORK SETTINGS

Configure the IP address by entering the “**Local network addresses address(0)**” parameter:

Network.address(x)		
Host IP address	192.168.22.41	<i>IP address of the IPefono</i>
Local network mask	255.255.0.0	<i>Mask of the IPefono</i>

FIGURE 27. CONFIGURATION OF THE IP ADDRESS

Take the following into account:

- ❑ The network settings (IP address, mask and Gateway) can be configured automatically from a DHCP server. Select the relevant parameter if you want to do this. When this option is selected, the IP address, router and name resolution (DNS) settings that have been configured will not apply.
- ❑ The device allows you to configure two IP addresses.

5.4 Setting the time

The time is set by first selecting the “**Time**” option (see [Figure 23](#)).

Time		
Greenwich Mean Time	1	<i>Time zone offset</i>
Network Time Server	0.0.0.0	<i>IP address of the clock synchronisation server</i>

FIGURE 28. TIME SETTING CONFIGURATION

5.5 Input and output ports

You configure the input and output ports by first selecting the “IO” option (see [Figure 23](#)).

Depending on the model of your IPefono device, you can configure a specific number of inputs and outputs.

IO

Modbus module address	1	<i>Address of the input and output module (information)</i>
TCP modbus port	0	<i>TCP port to control inputs and outputs</i>
UDP modbus port	0	<i>UDP port for control</i>
Input parameters	input(0)	<i>Parameters for input 1</i>
	input(1)	<i>Parameters for input 2</i>
	input(2)	<i>Parameters for input 3</i>
Output parameters	output(0)	<i>Parameters for output 1</i>
	output(1)	<i>Parameters for output 2</i>
Audio sensor parameters	audiosensor	<i>Audio sensor parameters</i>
Time in milliseconds to keep pressed before call	100	<i>Time during which you have to hold down the button before a call is made, in milliseconds</i>

FIGURE 29. CONFIGURATION OF INPUT AND OUTPUT PORTS

The IPefono can have up to three inputs and two outputs (depending on the model) which are identified with a numeric address. That address coincides with the range used in the MODBUS protocol, widely used in control and data acquisition systems. The inputs are numbered from the identifier 10000 and the outputs from 20000. This means that the inputs and outputs can be controlled by using applications that use the MODBUS protocol, without any further configuration required. The following table lists the identifier and the assigned port.

Model	NAME	IDENTIFIER	DESCRIPTION
HQ/LC	CALL	10000	Input for user call button
HQ	INPUT1	10001	Analogue or auxiliary digital input
	INPUT2	10002	Analogue or auxiliary digital input
HQ/LC	OUTPUT1	20000	Output to connect the device to open the door
HQ	OUTPUT2	20001	Auxiliary digital output

TABLE 1. ASSIGNING IDENTIFIERS FOR INPUTS AND OUTPUTS

Each input can be configured:

IO.input(x)		
Modbus address	10000	
Input name	CALL	<i>Input name</i>
Low input level is ON	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>This is activated when the input has low voltage</i>
Activate this output when ON	none ↓	<i>When the input is activated, the selected output is activated</i>
Play a message when this input is ON	_____	<i>When the input is activated, the specified file_name.wav is played</i>
Send this DTMF code when this input is activated and the call is established	_____	<i>Sends a DTMF code when the input is activated and the call is established</i>
It hangs up the current communication when it is activated	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Cuts off communications that are underway when the input is activated</i>

FIGURE 30. INPUT SETTINGS

Each output can be configured:

IO.output(x)		
Modbus address	20000	
Output name	OUT1	<i>Output name</i>
Normally closed	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Output normally open or closed</i>
Automatic deactivation time in milliseconds	1000	<i>When the output is activated, this is the time in milliseconds before it is automatically deactivated, '0' if it is deactivated manually</i>
Lock output meanwhile this input is ON	none ↓	<i>Locks the output if the input is activated</i>
Activate when the user type this code on the phone	*	<i>Key or dial to activate the output from the phone</i>
Deactivate when the user type this code on the phone	_____	<i>Key or dial to deactivate the output from the phone</i>
Activate when the user type this code in the keyboard	_____	<i>Activates the output when this code is keyed in on the keyboard</i>
Deactivate when the user type this code in the keyboard	_____	<i>Deactivates the output when this code is keyed in on the keyboard</i>
Activate it when the VoIP is connected	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Whether or not the output is only activated when in conversation</i>
Play a message when this output is activated	_____	<i>Plays a message when the output is activated</i>

FIGURE 31. OUTPUT SETTINGS

5.5.1 Audio sensor

The audio sensor is configured with the “**audiosensor**” settings (see [Figure 29](#)).

IO.audiosensor			
Audio sensitivity threshold	off	↓	<i>Minimum level for the audio sensor to be activated</i>
Time threshold in milliseconds	0		<i>How long the above level has to last, '0' if it should be activated immediately.</i>
Call to the destination when it is ON	<input type="radio"/> Yes	<input checked="" type="radio"/> No	<i>When the audio sensor is activated, the destination configured in SIP destination is called (VoIP/SIP settings)</i>
Play a message when it is ON	_____		<i>When the audio sensor is activated it plays file_name.wav</i>
Activate this output when it is ON	none	↓	<i>When the audio sensor is activated, the selected output is activated</i>

FIGURE 32. AUDIO SENSOR SETTINGS

Below there is an example of how to configure the audio sensor

Audio sensitivity threshold	<i>low</i>
Time threshold in milliseconds	<i>500</i>
Call to the destination when it is ON	<i>Yes</i>
Play a message when it is ON	<i>Alerta.wav</i>
Activate this output when it is ON	<i>OUT1</i>

FIGURE 33. AUDIO SENSOR EXAMPLE

5.6 Voice over IP settings

To configure the Voice over IP settings, first select the “**VoIP**” option (see [Figure 23](#)).

VoIP

Session Initiation Protocol	sip	<i>Configuration of the signalling protocol for SIP calls</i>
Automatically connect incoming calls	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>When the IPefono receives a call, the incoming call is automatically connected, otherwise the button has to be pressed to connect the call</i>
Connect incoming calls in listen mode by default	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>When the IPefono receives a call, it goes into listen mode (without connecting to the speaker).</i>
Wait time in seconds before connecting automatically the incoming calls	0	<i>Wait time before incoming calls are automatically answered</i>
Push the call button to talk	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Hold down the button to speak</i>
Use keyboard to call if it is available	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Use the keyboard to call</i>
Disable the call status light indicator	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Disable the call indicator light</i>
Echo canceller algorithm in a regular communication	attenuate↓	<i>Configuration of acoustic echo canceller</i>
Higher output level to cut the microphone input off	4000	
Echo canceller filter factor	8	
Echo canceller attenuation for the attenuate scheme	16	
Suppress the background noise	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Configure background noise suppressor</i>
Lower noise level allowed	10	
Conversation volume	medium↓	<i>Conversation volume control</i>
Ring volume	low ↓	<i>Ring volume control</i>
Tone volume	very low↓	<i>Tone volume control</i>
Pre-recorded messages volume	medium↓	<i>Volume of pre-recorded messages</i>
Auxiliary input volume	mute ↓	<i>Volume of the auxiliary line</i>
Emergency message volume	high ↓	<i>Emergency message volume</i>
Microphone input gain	very high↓	<i>Microphone gain</i>

Auxiliary input sensitivity	very low ↓	<i>Sensitivity of the auxiliary line to determine whether there is a signal at the input</i>
Real Time Protocol port	5004	<i>Real-time transport protocol port</i>
Voice codec for transmission	A-law ↓	<i>Audio encoder</i>
RTP packet length	20 ms ↓	<i>Length of the RTP packets</i>
RTP inactivity timeout in seconds	5	<i>Time before cutting off the call if no RTP packets are received</i>
Audio paging configuration	paging	<i>Configuration of the paging system</i>
Expiration time of the incoming call in second	120	<i>Maximum call time</i>
Expiration time of the outgoing call in second	120	<i>Maximum call time before proceeding to the alternative destination</i>
Expiration time of the conversations in minute	60	<i>Maximum conversation time</i>
Play information messages to the user	message	<i>Pre-recorded messages for the user</i>

FIGURE 34. CONFIGURATION OF VOIP SETTINGS

5.6.1 SIP protocol. VoIP connections

The configuration settings for the SIP protocol are as follows:

VoIP.sip		
Call button destination SIP URL	1@192.168.22.251	<i>Destination SIP address of the button</i>
Alternative SIP URL	_____	<i>Address if the previous one does not respond</i>
UDP port	5060	<i>UDP port of the SIP protocol</i>
STUN server	1000	<i>Address of the STUN server</i>
External IP address	_____	<i>IP address if it is static</i>
Proxy hostname or IP address	*	<i>IP address of the SIP VoIP provider</i>
Proxy account name	_____	<i>Account name of the VoIP provider</i>
Proxy account password	_____	<i>Registration password for the provider</i>
Domain name	_____	<i>The provider's domain name</i>
Proxy registration time in seconds	30	<i>How often the device is registered with the provider, in seconds</i>
Register only when an outgoing call is pending	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Only registered with the provider when a call has to be made</i>
Resolve the destination IP address using the called name	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Activates the automatic phonebook for shared dialling between IPefonos</i>

FIGURE 35. CONFIGURATION OF THE SIP PROTOCOL (VOIP OPTION)

Undoubtedly, the flexibility of SIP (Session Initiation Protocol) based telephony systems has been the key to the success of these systems, but it does require a minimum of knowledge to configure each of the environments that can be found. Listed below are the possibilities offered by intercoms, with details of how to configure the settings correctly in each case.

Bear in mind that SIP addresses are expressed in the following way:

- If the device is connected to a VoIP provider
 - sip:(name of the device being called)
- If it is not connected to a VoIP provider
 - sip:(name of the device being called)@(IP address of the device being called)

5.6.1.1 Point-to-point

This is the most basic mode. It consists of making calls between terminals, without any need for a PBX. It should be configured as follows:

Call button destination SIP URL	<i>4251@192.168.22.251</i>
Alternative SIP URL	
UDP port	<i>5060</i>
STUN server	
External IP address	<i>(not used)</i>
Proxy hostname or IP address	
Proxy account name	<i>(not used)</i>
Proxy account password	<i>(not used)</i>
Domain name	<i>(not used)</i>
Proxy registration time in seconds	<i>(not used)</i>
Register only when an outgoing call is pending	<i>No</i>
Resolve the destination IP address using the called name	<i>No</i>

FIGURE 36. POINT-TO-POINT CONNECTION CONFIGURATION (VOIP/SIP OPTION)

In each case, you have to specify the IP address and number that you want to call, using the nomenclature *<number called>@<IP address>* for the “**Call button destination SIP URL**” parameter.

In the “**Alternative SIP URL**” field you can add a second destination in case the first one does not reply or it is disconnected.

Normally the first part of the SIP URL is not used and the call rings at the terminal that has the destination IP, even if this number has not been configured or is not the same as the one dialled.

Some VoIP telephones do not work unless they have first been registered on a PBX. In those cases, simply specify that in the telephone’s settings that the PBX is one of our intercoms, this will allow the telephone to be registered and you will be able to operate it in point-to-point mode.

It is possible to work with numbers alone and not have to enter IP addresses for the SIP destinations, especially when calling from the telephone. To do this, you must activate the following option:

...	
Resolve the destination IP address using the called name	Yes

In this case, the number of the IPefono is the same as the name that has previously been configured on the “**Main**” screen (see [Figure 25](#)).

5.6.1.2 PBX

In slightly more complex environments, it may be useful to have a PBX so that you can offer more advanced services such as call queues, an automatic operator, call distribution etc.

Call button destination SIP URL	<i>4251</i>
Alternative SIP URL	
UDP port	<i>5060</i>
STUN server	
External IP address	
Proxy hostname or IP address	<i>192.168.22.10</i>
Proxy account name	<i>4041</i>
Proxy account password	<i>password</i>
Domain name	
Proxy registration time in seconds	<i>60</i>
Register only when an outgoing call is pending	<i>No</i>
Resolve the destination IP address using the called name	<i>No</i>

FIGURE 37. LOCAL SWITCHBOARD CONNECTION CONFIGURATION (VoIP/SIP OPTION)

In this environment we have to configure the IP address of the PBX in the “**Proxy hostname or IP address**” field, and the name and password configured for the IPefono on the PBX.

5.6.1.3 Internet VoIP services

You can also use internet VoIP services, which allow you to make calls to mobile phones, landlines or other IP destinations (after contracting the provider), without any adapter being required.

Call button destination SIP URL	<i>628777222</i>
Alternative SIP URL	
UDP port	<i>5060</i>
STUN server	<i>stun.l.google.com:19302</i>
External IP address	
Proxy hostname or IP address	<i>sip.adamvozip.es</i>
Proxy account name	<i>107107</i>
Proxy account password	<i>password</i>
Domain name	
Proxy registration time in seconds	<i>60</i>
Register only when an outgoing call is pending	<i>No</i>
Resolve the destination IP address using the called name	<i>No</i>

FIGURE 38. CONFIGURATION OF INTERNET VOIP SERVICES CONNECTION (VOIP/SIP OPTION)

So you simply have to enter the URL of the internet VoIP service provider (“**Proxy hostname or IP address**” settings), and the account name and password that it has assigned to you (“**Proxy account name**” and “**Proxy account password**” settings). Also remember to configure the DNS and Gateway on the “**Network**” screen)

...
Router IP address	0.0.0.0	<i>Address of the router (Gateway)</i>
Domain name server	0.0.0.0	<i>Name resolution IP address (DNS)</i>
...

FIGURE 39. CONFIGURATION OF INTERNET VOIP SERVICES CONNECTION (NETWORK OPTION)

As you are making calls over the public network, the IP addresses that are used for SIP signalling will not be the same as those used by the network and in some cases the router IP will not be able to route the packets to the destination in question. STUN servers are used to resolve this problem; they provide the public IP address to the IPefono so that it can be used in signalling rather than its local IP address (“**STUN server**” settings). Finally, if you are connected to the internet via a static IP address, you can specify this in the “**External IP address**” field and dispense with the STUN server.

5.6.1.4 Gateway to the telephone network

In a point-to-point environment you can use a VoIP gateway to divert calls from IPefonos to mobile phones or landlines.

Call button destination SIP URL	<i>628777222@192.168.22.42</i>
Alternative SIP URL	
UDP port	<i>5060</i>
STUN server	
External IP address	
Proxy hostname or IP address	
Proxy account name	
Proxy account password	
Domain name	
Proxy registration time in seconds	<i>(not used)</i>
Register only when an outgoing call is pending	<i>No</i>
Resolve the destination IP address using the called name	<i>No</i>

FIGURE 40. CONFIGURATION OF THE GATEWAY'S CONNECTION TO THE TELEPHONE NETWORK (VOIP/SIP OPTION)

To do this you have to configure the adapter accordingly and, in the destination field, specify the number called, according to the following format: <telephone number>@<the Gateway's IP address>.

5.6.2 Acoustic echo canceller

One of the main disadvantages of “hands free” communications systems is that there is an echo on the remote terminal. With traditional communications systems, this echo has always occurred on the transition from 2 to 4 wires, but it did not cause any problems for the speaker as there was no delay. To the contrary, it told the speaker that the line was working properly.

With the introduction of voice communications over switched networks, and subsequently over IP networks, this delay has increased considerably, even going so far as to hinder proper communication between both points. It therefore became necessary to cancel the echo, so that the speaker does not hear him/herself speak. This obviously removed one of the major advantages of traditional communications, the confirmation that the speaker’s voice is being heard at the other end.

It is worth noting that the acoustic echo cancellers that are normally used are usually prepared for a “clean” type of environment, i.e. when the speaker talks, the return audio comes from his/her voice, with minimal distortion, plus the voices speaking on the other end. However, when the speaker’s voice becomes too distorted (either due to the location of the microphone and speaker or the volume being too high), the adaptive cancellation algorithm stops working properly and introduces noises, especially at the beginning and end of sentences.

Given that IPefonos are designed to work in very diverse environments, they incorporate a variety of echo cancellation mechanisms, so they can be optimised according to the possibilities offered by the environment. The different alternatives that are available are listed below.

Naturally, before configuring the echo canceller, it is essential to set the conversation volume that you are going to work with, otherwise, when this is changed, it may stop working properly.

Conversation volume	medium↓	<i>Set the conversation volume control</i>
---------------------	---------	--

FIGURE 41. CONFIGURATION OF THE ACOUSTIC ECHO CANCELLER (VOIP OPTION)

5.6.2.1 Adaptive echo cancellation

The adaptive echo canceller is configured in the Voice over IP settings (“**VoIP**” option) with the following values:

Echo canceller algorithm in a regular communication	<i>suppression</i>
Higher output level to cut the microphone input off	<i>(not used)</i>
Echo canceller filter factor	<i>(not used)</i>
Echo canceller attenuation for the attenuate scheme	<i>4</i>

FIGURE 42. CONFIGURATION OF THE ADAPTIVE ECHO CANCELLER (VOIP OPTION)

This is the standard cancellation mode and it consists of removing the return audio that comes through the microphone, the audio emitted by the speaker, but with a time delay. For this algorithm to work you have to continuously calculate the delay that needs to be added to the output audio so that the right amount can be removed, as well as the attenuation that will be added to the signal before it is removed. In order to calculate these values, the system needs to be capable of identifying what was emitted within the return signal, which is rarely possible in industrial or noisy environments such as car parks, airports, factories etc.

You can test if it works and, if not, you can change the algorithm by added “extra” attenuation (“**Echo canceller attenuation for the attenuate scheme**” settings, with values such as 2, 4, 8 or 16). If you do not achieve the desired effect you should try with another cancellation model.

5.6.2.2 Echo cancellation through attenuation

Echo cancellation through attenuation is configured in the Voice over IP settings (“VoIP” option) with the following values:

Echo canceller algorithm in a regular communication	<i>attenuate</i>
Higher output level to cut the microphone input off	<i>4000</i>
Echo canceller filter factor	<i>8</i>
Echo canceller attenuation for the attenuate scheme	<i>16</i>

FIGURE 43. CONFIGURATION OF ECHO CANCELLATION THROUGH ATTENUATION (VOIP OPTION)

If the system is not able to recognise the echo from the return signal, you can choose to attenuate what is heard while you speak, to minimise the uncomfortable effect of hearing yourself while you speak. There are two reasons for attenuating it and not completely eliminating it. On the one hand, you can tell if the speaker on the other end is talking at the same time as you and, on the other, you can check if the remote system is working properly, what it is emitting and you can hear what is being said.

You need to change three settings in this mode. The first, “Higher output level to cut the microphone input off” (*4000*) indicates the level from which the IPefono interprets that is being emitted by the speaker and will subsequently mute the microphone signal. You should test the values, from lowest to highest (2000 to 8000).

The second parameter, “Echo canceller filter factor” (*8*), indicates how long the intercom will continue to cut off the microphone signal after determining that a voice is no longer being emitted. This setting is designed to prevent “mini-outages” while the speaker is saying a sentence, but it should also be sufficiently low to switch to normal mode when he/she has finished speaking. Setting it to 8 normally has good results.

The final one is the echo attenuation factor, “**canceller attenuation for the attenuate scheme**” (*16*), which indicates the value by which the return signal is divided while the speaker talks. It will depend directly on the volume that has been set and the acoustics that you have.

5.6.2.3 Half-duplex echo cancellation

Half-duplex cancellation is configured in the Voice over IP settings (“VoIP” option) with the following values:

Echo canceller algorithm in a regular communication	<i>mute</i>
Higher output level to cut the microphone input off	<i>4000</i>
Echo canceller filter factor	<i>8</i>
Echo canceller attenuation for the attenuate scheme	<i>(not used)</i>

FIGURE 44. CONFIGURATION OF HALF-DUPLEX ECHO CANCELLATION (VOIP OPTION)

There are extreme cases where cancellation through attenuation is not enough and you simply want to completely eliminate the return signal during speaking. For example, when a Voice over IP gateway is used to an analogue telephone, there is a closed loop in the audio signal. The telephone line, as indicated above, has an echo on it, and you also have a closed loop in the microphone-speaker set. This can result in signal coupling, so the system stops working altogether. If this is the case, you need to work in this mode.

For the speaker it is a “half-duplex” mode, because he/she will not hear anything while he/she speaks. For the intercom, the communication will be “full-duplex”.

You need to change two settings in this mode. The first, **“Higher output level to cut the microphone input off” (4000)** indicates the level from which the IPefono interprets that is being emitted by the speaker and will subsequently mute the microphone signal. As before, you should test the values, from lowest to highest (2000 to 8000).

The second parameter, “Echo canceller filter factor” (8), indicates how long the intercom will continue to cut off the microphone signal after determining that a voice is no longer being emitted. This setting is designed to prevent “mini-outages” while the speaker is saying a sentence, but it should also be sufficiently low to switch to normal mode when he/she has finished speaking. Setting it to 8 normally has good results.

5.6.2.4 No echo cancellation

Echo canceller algorithm in a regular communication	<i>disabled</i>
Higher output level to cut the microphone input off	<i>(not used)</i>
Echo canceller filter factor	<i>(not used)</i>
Echo canceller attenuation for the attenuate scheme	<i>(not used)</i>
...	
RTP packet length	<i>20 ms</i>

FIGURE 45. CONFIGURATION WITHOUT ECHO CANCELLATION (VOIP OPTION)

When you only work on a local network, you can dispense with echo cancellation because it is not considered to be a problem when the time that elapses between you speaking and hearing yourself is less than 100 milliseconds. Therefore, you only have to bear in mind that the size of the packets should be as small as possible, because that is where the delay occurs. You should also check the configuration of the VoIP telephones in your facility and minimise the configuration settings where “jitter-buffer” is specified, to minimise the delay introduced by these terminals as well.

5.6.3 Background noise canceller

The IPefono can attenuate the signal when the speaker is not talking, so the background noise that occurs is reduced and it does not bother the listener.

To do this, the following settings need to be configured:

Suppress the background noise	Yes
Lower noise level allowed	10

FIGURE 46. CONFIGURATION OF THE BACKGROUND NOISE CANCELLER (VOIP OPTION)

One possible initial value for the “**Lower noise level allowed**” setting is 10. If you still hear background noise, this must be increased. If you cannot even hear the speaker, this setting must be reduced. Normal values are between 0 and 100.

5.6.4 Paging

One of IPefono’s additional features is the possibility of “*paging*” several intercoms simultaneously from the control centre, either to different zones or to all of them at the same time.

Paging is configured using the “**VoIP/paging**” option (see [Figure 34](#)), the settings are as follows:

VoIP.paging		
IP address for audio paging	224.192.0.17	<i>Multicast IP address to which the paging messages are sent</i>
RTP port for audio paging	5004	<i>RTP port for paging</i>
Incoming group paging zone	anyone ↓	<i>Paging group to which the device belongs</i>
Broadcast the audio received in the auxiliary input	● Yes ○ No	<i>Retransmits the audio received by the auxiliary line</i>
Voice codec for transmission	u-Law ↓	<i>Voice codec for the transmission</i>
Group paging zone to transmit the auxiliary signal	all ↓	<i>Zone to which the auxiliary line is retransmitted</i>

FIGURE 47. CONFIGURATION OF THE PAGING SYSTEM (VOIP/PAGING OPTION)

For this, you have to select a multicast IP address and the RTP port that will be used for paging, as well as specifying the IPefono group or zone.

From the remote terminal you will have to establish RTP communication with the selected multicast address. Some VoIP telephones on the market, such as those manufactured by “Yealink”, allow you to configure multicast RTP communications. Another option is to do it with your own software.

5.6.5 Messages

Messages should be WAV files and in the configuration settings you should indicate the filenames of the messages with their extensions:

VoIP.message		
Calling to the Helpdesk	_____	<i>Message when starting a call</i>
Incoming call notification	_____	<i>Message when you receive an incoming call</i>
Incoming call connected	_____	<i>Message when connecting the call</i>
No response from called	_____	<i>Message when there is no response from the destination</i>
Out of Service	_____	<i>Out of service message</i>
A timeout finished the call	_____	<i>Message when the IPefono stops calling after its internal timer has expired</i>

FIGURE 48. CONFIGURATION OF USER MESSAGES (VOIP/PAGING OPTION)

5.7 Video settings

Video settings are configured by first selecting the “**Video**” option (see [Figure 23](#)) and they are applied if the IPefono has a video camera.

Video

Camera model	C429 ↓	<i>Model of the camera</i>
Image resolution	QVGA ↓	<i>Resolution of the image</i>
Image compression. Frameratevs quality	high ↓	<i>Image compression</i>
Mirror image	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Show mirror image</i>
Force indoor profile	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Image profile for indoors</i>
Backlight compensation	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Back light compensation</i>

FIGURE 49. CONFIGURATION OF THE VIDEO SETTINGS

Therefore, you have to select the camera that has been connected, the resolution – bearing in mind that the higher the resolution the lower the image refresh rate – and what compression to use.

When the video has been configured, you need to save the configuration and restart the IPefono for it to take effect. To see if it is working properly, simply enter the following URL into your browser:

<http://192.168.22.41/webcam.html> or <http://192.168.22.41/frame.jpg>

IPefonos are not used as IP video phones and they are not compatible with them. They are video intercom systems and they are designed to enable you to view the camera display irrespective of the call state.

It is therefore necessary to have software that integrates video. ConectaIP has two applications. On the one hand, *IPefono Helpdesk*, designed for Windows and which requires a PC to work and, on the other hand, Video IPefono for Android, which specifically needs the Grandstream GXP2200 or GXV3240 VoIP phone.



FIGURE 50. ANDROID APPLICATION ON THE VOIP TELEPHONE

To use the Android software, you first need to update the firmware for the GXP2200/GXV3240 telephone to its latest version and then install an APK file from a pen drive using the “File Manager” that comes with the phone.

Once installed, you have to set up an “intercom” account so that it can be registered with any of the intercoms in your facility.

Account active :	<i>Yes</i>
Account name :	<i>Intercom</i>
SIP server :	<i>192.168.22.41</i>
SIP User ID :	<i>4001</i>
Authenticate ID :	<i>4001</i>
Authenticate password :	<i>4001</i>

Finally, the phonebook is configured with all of the IPefonos in your facility. In the “*phone*” field you have to enter the IP address of the intercom (*192.168.22.41*), the “intercom” account name that you have created and, below, in “*WEB homepage*” the URL for the camera’s images (<http://192.168.22.41/frame.jpg>).

When configuration is complete, you can open the “*Video IPefono*” application. It will connect to the camera of the intercom that is currently selected and you will be able to establish communication or activate its outputs from the touchscreen.

5.8 Keyboard settings

The IPefono's electronics allow you to connect a matrix keyboard with four rows and four columns, so it has an additional 16 keys.

There are several functions that can be executed from the keyboard, each of which can be configured from the “**Keyboard**” configuration screen (see [Figure 23](#)).



Keyboard		
Use keyboard to call	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Use the keyboard to call</i>
Use keyboard to activate relays	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Use to activate outputs</i>
Beep the speaker when the key is pressed	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Beep when the key is pressed</i>
Key to cancel the current call	none ↓	<i>Key to cancel the call</i>
Key to cancel the current call	none ↓	<i>End of dialling key</i>
Timeout to call to the current dial in seconds	0	<i>Timeout before starting dialling</i>
SIP destination when the 'A' key is pressed	_____	<i>SIP address of the 'A' key</i>
SIP destination when the 'B' key is pressed	_____	<i>SIP address of the 'B' key</i>
SIP destination when the 'C' key is pressed	_____	<i>SIP address of the 'C' key</i>
SIP destination when the 'D' key is pressed	_____	<i>SIP address of the 'D' key</i>
Play a message when the '0' key is pressed	_____	<i>Message when pressing the '0' key</i>
Play a message when the '1' key is pressed	_____	
Play a message when the '2' key is pressed	_____	
Play a message when the '3' key is pressed	_____	
Play a message when the '4' key is pressed	_____	
Play a message when the '5' key is pressed	_____	
Play a message when the '6' key is pressed	_____	
Play a message when the '7' key is pressed	_____	
Play a message when the '8' key is pressed	_____	
Play a message when the '9' key is pressed	_____	
Play a message when the 'A' key is pressed	_____	
Play a message when the 'B' key is pressed	_____	
Play a message when the 'C' key is pressed	_____	
Play a message when the 'D' key is pressed	_____	

FIGURE 51. CONFIGURATION OF KEYBOARD SETTINGS

Below there is an example of how to configure the keyboard settings:

Use keyboard to call	Yes
Use keyboard to activate relays	Yes
Beep the speaker when the key is pressed	Yes
Key to cancel the current call	#
Key to cancel the current call	*
Timeout to call to the current dial in seconds	2
SIP destination when the 'A' key is pressed	4041@192.168.22.41
SIP destination when the 'B' key is pressed	4042@192.168.22.42
SIP destination when the 'C' key is pressed	4043@192.168.22.43
SIP destination when the 'D' key is pressed	
Play a message when the "0" key is pressed	
...	
Play a message when the 'D' key is pressed	OutofService.wav

FIGURE 52. EXAMPLE OF CONFIGURATION OF THE KEYBOARD SETTINGS

5.8.1 Making calls

If you activate the option to use the keyboard for calling, the IPefono WMT can work in two different ways:

- ❑ You can directly dial the extension number that you want to reach. If you have configured a SIP proxy (see section [5.6.1 SIP protocol. VoIP connections](#)), the number dialled is sent to the proxy so that it can determine the destination.
- ❑ If you are using a PBX (you have not configured a SIP proxy), it will be like dialling on a VoIP telephone, when using a point-to-point scenario the IPefono will call the following SIP address:

sip://<number dialled>@<subnet address>.<number dialled>

So, for example, if your IP address is 192.168.22.41, the mask is 255.255.255.0 and you dial the number 123, the intercom will call the SIP URL *sip://123@192.168.22.123*.

When you dial with the keyboard, the IPefono will determine that dialling has finished either because the interval set between each digit has been reached or because the key that has been set for that purpose has been pressed (# by default).

5.8.2 Activating outputs

If you enable the option “**Use keyboard to activate relays**”, you can activate the intercom's outputs by using the keyboard to dial the code that has been set in the “**Activate when the user type this code**” settings on the output configuration screen (see [Figure 31](#)).

5.8.3 Playing messages

In settings you can enter the name of an audio file so that, when the set key is pressed, the intercom will play the saved audio file over the speaker. These files should be stored in the IPefono's internal storage unit.

5.8.4 Activating emergency mode

Finally, you can activate or deactivate the intercom's emergency mode by entering the set code on the **Emergency** screen (see [Figure 53](#)).

5.9 Emergency Mode Settings

IPefonos have an emergency status, during which they continually repeat a pre-recorded message and tone. You can configure the message that is played, the tone between messages and the time that passes before the message is repeated again.

You can also program, when it enters emergency mode, that some of the outputs are automatically activated to signal the event to other devices.

To configure the Emergency Mode settings, first select the **“Emergency”** option (see [Figure 23](#)).

Emergency		
Input to activate the emergency mode	none ↓	<i>Input that should be activated to enter into Emergency mode</i>
Activate this mode when user type this code	_____	<i>Activate Emergency mode when this code is entered on the keyboard</i>
Deactivate it when the user type this code	_____	<i>Deactivate with this code</i>
Play this message when the device is in emergency mode	_____	<i>Play the message when it is in this mode</i>
Time before replaying the message (seconds)	30	<i>Time between messages</i>
The emergency mode will be disabled automatically when this time (sec) is reached	360	<i>Automatically deactivate when the specified time is reached</i>
Tone for playing between messages	silence ↓	<i>Tone between messages</i>
Activate this output in emergency mode	none ↓	<i>Activate selected output</i>
Broadcast the audio played in this mode to other devices on the same paging zone	<input type="radio"/> Yes <input checked="" type="radio"/> No	<i>Broadcast the audio to the other intercoms in the same zone</i>

FIGURE 53. CONFIGURATION OF THE EMERGENCY MODE SETTINGS

There is an example of how to configure it below:

Input to activate the emergency mode	<i>IN1</i>
Activate this mode when user type this code	<i>*701</i>
Deactivate it when the user type this code	<i>*700</i>
Play this message when the device is in emergency mode	<i>Danger.wav</i>
Time before replaying the message in seconds	<i>30</i>
The emergency mode will be disabled automatically when this time (in seconds) is reached	<i>360</i>
Tone for playing between messages	<i>ring</i>
Activate this output in emergency mode	<i>OUT1</i>
Broadcast the audio played in this mode to other devices on the same paging zone	<i>Yes</i>

FIGURE 54. EXAMPLE OF CONFIGURATION OF THE EMERGENCY MODE

The IPefono will go into emergency mode under any of the following conditions:

- ❑ The configured input signal is activated (in the example *IN1*).
- ❑ The emergency code is entered on the keyboard or a remote terminal (**701*).
- ❑ The “EmergencyStart” HTTP command is used
(<http://192.168.22.41/EmergencyStart>).

And it will exit that mode under any of the following conditions:

- ❑ The configured input signal is deactivated, if this is the cause of the emergency.
- ❑ The end of emergency code is entered on the keyboard or a remote terminal (**700*).
- ❑ The “EmergencyStop” HTTP command is used
(<http://192.168.22.41/EmergencyStop>).
- ❑ The maximum time that the IPefono can remain in this mode has passed (*360*).

5.10 Auto-configuration

If your facility has several IPefonos, the auto-configuration feature makes it easier to start each of them up.

These are the steps to follow:

If the first IPefono has not been shipped with the necessary default settings, configure the network settings by selecting the “*Network*” option (see [5.3 Network configuration](#)) and the Voice over IP settings in the “*VoIP*” option (see [5.6 Voice over IP settings](#)).

This first IPefono is connected to the network and started up. Hold down the device’s call button or the Factory button (see its location in [Figure 4 IPefono HQ connections](#)) for at least 10 seconds.

Then the second IPefono is also connected to the network and the call button or the Factory button is held down for at least 10 seconds. The device’s LED light flashes and it is automatically configured with the following IP address and the following name (“*Device name*” settings on the “*Main*” screen). The rest of the “*Network*” and “*VoIP*” settings are copied from the first device.

The third IPefono is connected to the network and the previous step is repeated.

And so on.

The auto-configuration function only runs the first time that the button is held down for at least 10 seconds, it will have no effect on subsequent times. If you want to reactivate this feature on the IPefono, enter the configuration and change the “*Auto set up is activated*” setting (see [Figure 25. Configuration of general settings](#)).

6

System administration

6.1 Entering system administration

On all of the device's configuration screens you can enter system administration via the **"tools"** hyperlink or the **"System tools"** option on the home screen (see [Figure 6](#)).

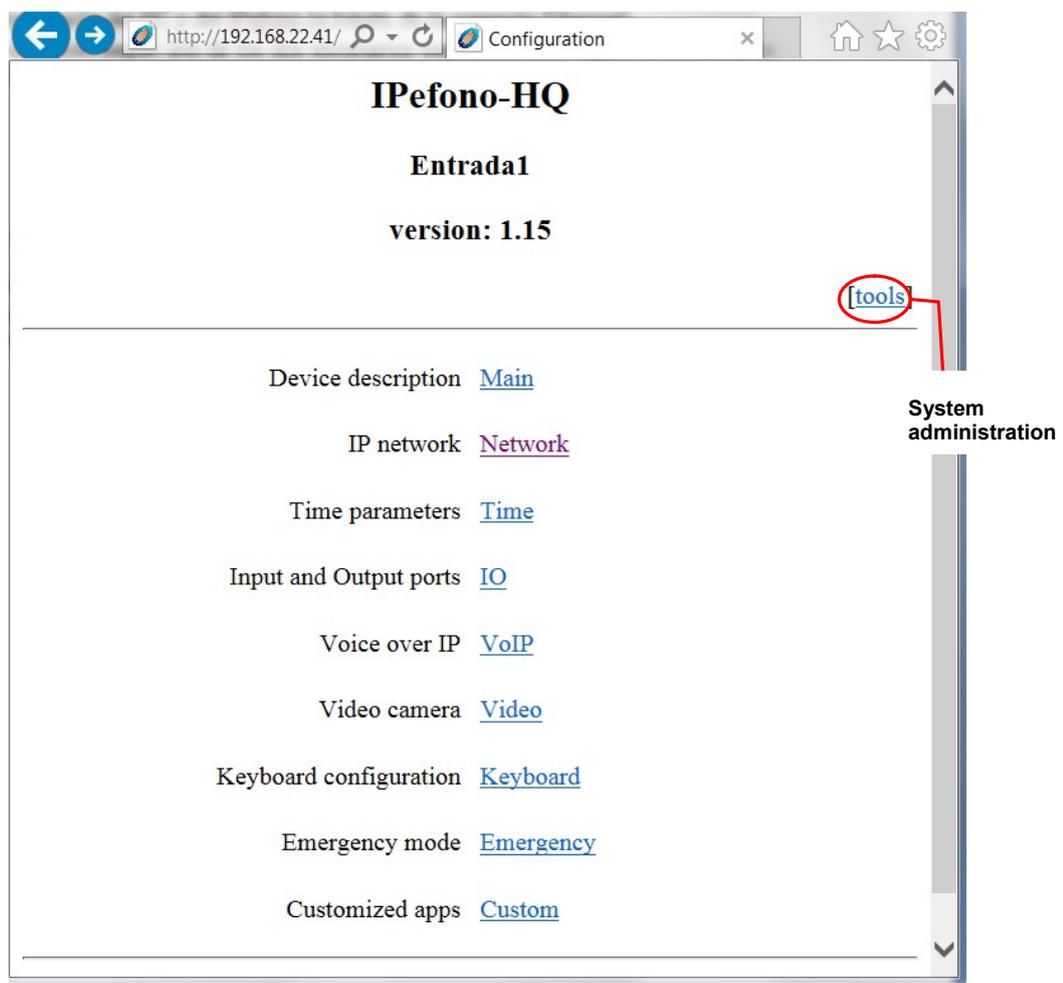


FIGURE 55. ENTERING SYSTEM ADMINISTRATION

6.2 Administration options

The system administration options are as follows:

Tools

List configuration	<i>List configuration</i>
Save configuration	<i>Save configuration</i>
Save and apply configuration	<i>Save and apply configuration</i>
Undo changes	<i>Undo changes</i>
Download configuration	<i>Download configuration</i>
Upload file	<i>Upload file</i>
Upload configuration	<i>Upload configuration</i>
Update firmware	<i>Update the firmware</i>
Status report	<i>Registration status</i>
Reset device	<i>Reset device</i>

FIGURE 56. SYSTEM ADMINISTRATION OPTIONS OF THE TOOLS HYPERLINK

6.3 Configuration file management

6.3.1 Saving the configuration

The “**Save configuration**” option shown in [Figure 56](#) allows you to save the configuration of the device on which you are working. This will take effect when it restarts (“**Reset device**” option in [Figure 56](#)). If you want to save and apply it at the time, select the “**Save and apply configuration**” option shown in [Figure 56](#).

6.3.2 Configuration list

The configuration file is a ‘config.ini’ text file and, as such, it can be edited, changed and resent.

The “**List configuration**” option shown in [Figure 56](#) allows you to display the device’s settings. Below there is an example of a configuration file, indicating the respective settings:

Main.type= IPefono-HQ	Device type
Main.version = 1.15	Firmware version
Main.name = 4041	Device name
Main.description= <i>Entrada1</i>	Device description
Main.contact =	Administrator contact
Main.location =	Device location
Main.username =	Administrator username
Main.password =	Administrator password
Network.bAutoConfiguration = <i>FALSE</i>	Get network configuration from DHCP server
Network.address(0).ip = <i>192.168.22.41</i>	Host IP address
Network.address(0).mask = <i>255.255.0.0</i>	Local network mask
Network.ipGateway = <i>0.0.0.0</i>	Router IP address
Network.ipDnsServer = <i>0.0.0.0</i>	Domain name server
Network.domainName = <i>CONECTAIP</i>	Domain name
Network.portTelnet = <i>23</i>	Telnet server port
Network.portWeb = <i>80</i>	Web server port
Network.portDNS = <i>53</i>	Domain Name Server port
Network.portNTP = <i>123</i>	Network Time Protocol port
Network.portSNMP = <i>161</i>	Simple Network Management Protocol port
Network.ipSnmpTrapDestination =	SNMP trap URL destination
Network.portTrapSNMP = <i>162</i>	SNMP trap port destination
Network.idVLAN = <i>0</i>	Virtual Network identifier (2-4095,0=disabled)
Network.priorityVLAN = <i>0</i>	Virtual Network priority (0-7)

Time.GMT = 1	Greenwich Mean Time
Time.ipSntpServer = 0.0.0.0	Network Time Server
IO.address = 1	Modbus module address
IO.portTcp = 0	TCP modbus port
IO.portUdp = 0	UDP modbus port
IO.input(0).address = 10000	Modbus address
IO.input(0).szName = CALL	Input name
IO.input(0).inverted = FALSE	Low input level is ON
IO.input(0).output = none	Activate this output when ON
IO.input(0).speaker =	Play a message when this input is ON
IO.output(0).address = 20000	Modbus address
IO.output(0).szName = OUT1	Output name
IO.output(0).inverted = FALSE	Normally closed
IO.output(0).autoOffTime = 2000	Automatic deactivation time in milliseconds
IO.output(0).input = none	Lock output meanwhile this input is ON
IO.output(0).keyCodeOn = *	Activate when the user type this code
IO.output(0).keyCodeOff =	Deactivate when the user type this code
IO.output(0).useLocalKeyboard = FALSE	Use local keyboard to control this output
IO.output(0).activeWhenConnected = FALSE	Activate it when the VoIP is connected
IO.audioSensor.level = off	Audio sensitivity threshold
IO.audioSensor.time = 0	Time threshold in milliseconds
IO.audioSensor.call = FALSE	Call to the destination when it is ON
IO.audioSensor.speaker =	Play a message when it is ON
IO.audioSensor.output = none	Activate this output when it is ON
IO.callButtonFilter = 100	Time in milliseconds to keep pressed ...
VoIP.sip.destination = 4251@192.168.22.251	Call button destination SIP URL
VoIP.sip.backup =	Alternative SIP URL
VoIP.sip.port = 5060	UDP port
VoIP.sip.stun =	STUN server
VoIP.sip.externallIP =	External IP address
VoIP.sip.proxy =	Proxy hostname or IP address
VoIP.sip.phoneName =	Proxy account name
VoIP.sip.password =	Proxy account password
VoIP.sip.domain =	Domain name

VoIP.sip.registerTime = 30	Proxy registration time in seconds
VoIP.sip.registerOnCall = <i>FALSE</i>	Register only when an outgoing call is pending
VoIP.sip.autoResolveName = <i>FALSE</i>	Resolve the destination IP address using ...
VoIP.autoconnect = <i>TRUE</i>	Automatically connecting coming calls
VoIP.listenModelIncommingCall = <i>FALSE</i>	Connect incoming calls in listen mode ...
VoIP.autoconnectTime = 0	Wait time in seconds before connecting ...
VoIP.pushToTalk = <i>FALSE</i>	Push the call button to talk
VoIP.useKeyboard = <i>FALSE</i>	Use keyboard to call if it is available
VoIP.disableLightSignal = <i>FALSE</i>	Disable the call status light indicator
VoIP.echoCanceller = <i>attenuate</i>	Echo canceller algorithm in a regular ...
VoIP.echoCancellerLevel = 4000	Higher output level to cut the microphone ...
VoIP.echoCancellerFilter = 8	Echo canceller filter factor
VoIP.echoAttenuation = 16	Echo canceller attenuation for the ...
VoIP.noiseSuppression = <i>FALSE</i>	Suppress the back ground noise
VoIP.noiseSuppressionLevel = 10	Lower noise level allowed
VoIP.volume = <i>medium</i>	Conversation volume
VoIP.volumeRing = <i>low</i>	Ring volume
VoIP.volumeTone = <i>verylow</i>	Tone volume
VoIP.volumeMessage = <i>medium</i>	Pre-recorded messages volume
VoIP.volumeAux = <i>mute</i>	Auxiliary input volume
VoIP.volumeEmergency = <i>high</i>	Emergency message volume
VoIP.gainMic = <i>veryhigh</i>	Microphone input gain
VoIP.sensitivityAux = <i>verylow</i>	Auxiliary input sensitivity
VoIP.portRTP = 5004	Real Time Protocol port
VoIP.codec = <i>A-Law</i>	Voice codec for transmission
VoIP.packetTime = 40 ms	Rtp packet length
VoIP.rtpInactivityTimeout = 5	RTP inactivity timeout in seconds
VoIP.paging.ipMulticast = 224.192.0.17	IP address for audio paging
VoIP.paging.portMulticast = 5004	RTP port for audio paging
VoIP.paging.group = <i>all</i>	Group paging address
VoIP.maximumRingDuration = 120	Expiration time of the incoming call in ...
VoIP.maximumOutgoingCallDuration = 120	Expiration time of the outgoing call in ...
VoIP.maximumCallDuration = 60	Expiration time of the conversations in ...

VoIP.message.calling =	Calling to the Helpdesk
VoIP.message.proceeding =	Incoming call notification
VoIP.message.connect =	Incoming call connected
VoIP.message.noResponse =	No response from called
VoIP.message.outOfService =	Out of Service
VoIP.message.timeout =	A timeout finished the call
VoIP.message.emergency =	Emergency call
Video.cameraModel = C429	Image resolution
Video.resolution = QVGA	Image compression. Frame rate vs quality ...
Video.compression = <i>high</i>	Camera model
Keyboard.useToCall = <i>FALSE</i>	Use keyboard to call
Keyboard.useToActivateRelays = <i>FALSE</i>	Use keyboard to activate relays
Keyboard.beepOnPressKey = <i>FALSE</i>	Beep the speaker when the key is pressed
Keyboard.cancelKey = <i>none</i>	Key to cancel the current call
Keyboard.dialKey = <i>none</i>	Key to call to the current dial
Keyboard.dialTimeout = 0	Timeout to call to the current dial in seconds
Keyboard.directDialA =	SIP destination when the 'A' key is pressed
Keyboard.directDialB =	SIP destination when the 'B' key is pressed
Keyboard.directDialC =	SIP destination when the 'C' key is pressed
Keyboard.directDialD =	SIP destination when the 'D' key is pressed
Keyboard.speaker0 =	Play a message when the '0' key is pressed
Keyboard.speaker1 =	Play a message when the '1' key is pressed
Keyboard.speaker2 =	Play a message when the '2' key is pressed
Keyboard.speaker3 =	Play a message when the '3' key is pressed
Keyboard.speaker4 =	Play a message when the '4' key is pressed
Keyboard.speaker5 =	Play a message when the '5' key is pressed
Keyboard.speaker6 =	Play a message when the '6' key is pressed
Keyboard.speaker7 =	Play a message when the '7' key is pressed
Keyboard.speaker8 =	Play a message when the '8' key is pressed
Keyboard.speaker9 =	Play a message when the '9' key is pressed
Keyboard.speakerA =	Play a message when the 'A' key is pressed
Keyboard.speakerB =	Play a message when the 'B' key is pressed
Keyboard.speakerC =	Play a message when the 'C' key is pressed

Keyboard.speakerD =	Play a message when the 'D' key is pressed
Emergency.input = <i>none</i>	Input to activate the emergency mode
Emergency.keyCodeOn =	Activate this mode when user type this ...
Emergency.keyCodeOff =	Deactivate it when the user type this code
Emergency.emergencyMessage =	Play this message when the device is in ...
Emergency.pauseTime = 30	Time before replaying the message in ...
Emergency.expirationTime = 360	The emergency mode will be disabled ...
Emergency.tone = silence	Tone for playing between messages
Emergency.output = none	Activate this output in emergency mode

FIGURE 57. IPEFONO CONFIGURATION TEXT FILE

6.3.3 Editing the configuration file

The configuration file can be downloaded from the device by using the “**Download configuration**” option shown in [Figure 56](#). Once it is on your PC it can be edited with any text editor and changed while retaining the original format. Use the “**Upload configuration**” option to upload the new configuration. The system has to be restarted for it to take effect, by selecting the “**Reset device**” option shown in [Figure 56](#).

6.4 Updating firmware

Updating the version of the device is a very simple process. Select the “**Update firmware**” option from the “**Tools**” hyperlink.

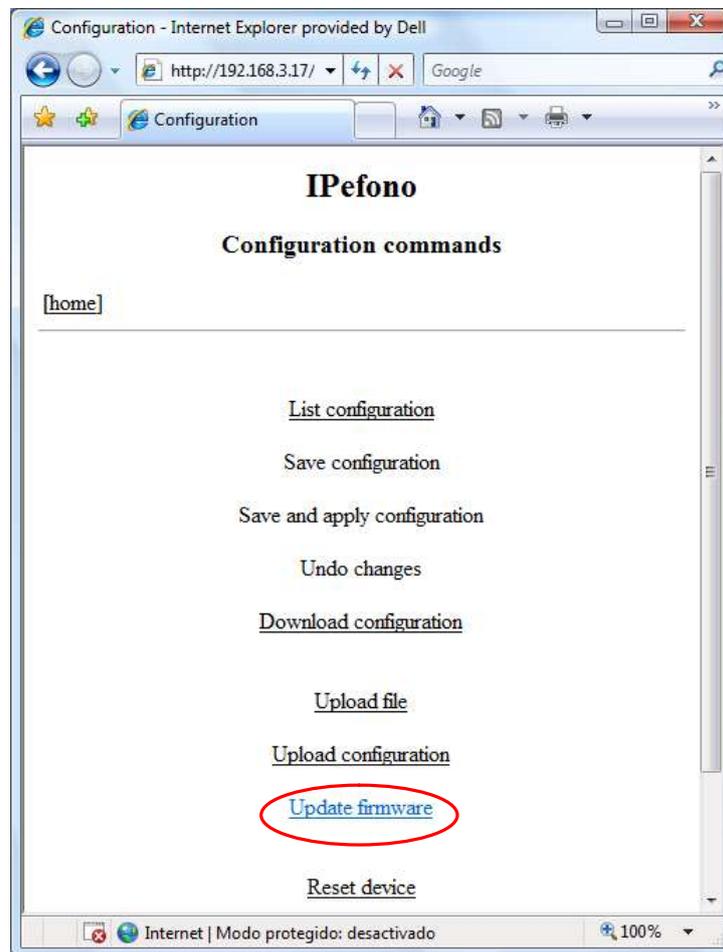


FIGURE 58. UPDATING FIRMWARE

When you do this, a new window appears on which the **Browse...** button allows you to select the location of the new firmware version on your PC.

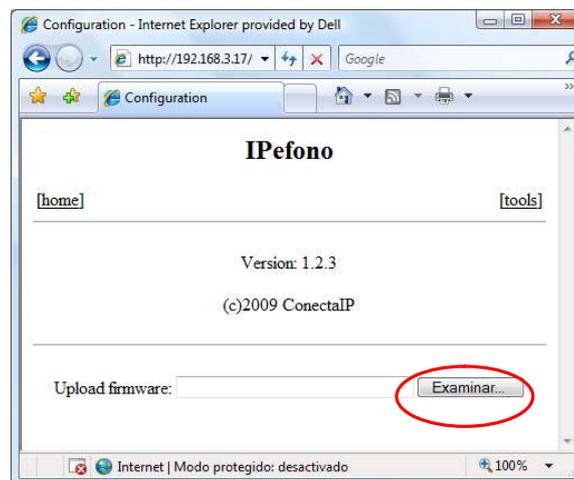


FIGURE 59. FIRMWARE SELECTION

The firmware is protected by a digital signature, so it does not accept incorrect or modified files. The system notifies you if an invalid file is loaded.

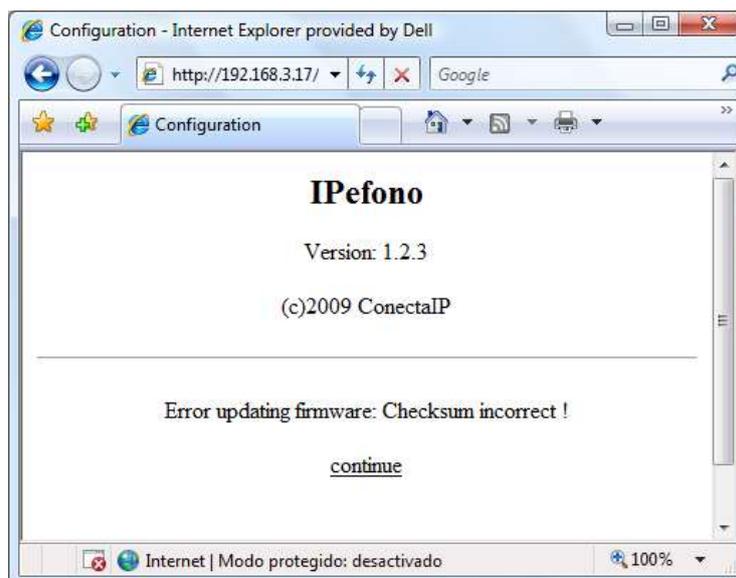


FIGURE 60. INCORRECT FIRMWARE UPDATE

If it has been successfully updated a new window appears, showing the current version and information about the new firmware: version, time and date the version was created and build number. The user is then presented with two options: perform the update or cancel it.

6.5 Estado del dispositivo

You can view the status of the device by selecting the option “**Status report**” from the “**Tools**” hyperlink (see Figure 56) or “**System tools**” option on the home screen (see Figure 6).

Device Status

[refresh]

Name	Status
Start up time	0 days, 3 hours, 29 minutes, 28 seconds
CPU workload	0 %
Free memory	19232 bytes
SIP server registration	not configured
Communication	idle [call]
Last call destination	
Last audio test result	fail [test]
Emergency situation	off [activate]
Video camera	not detected
input(0) = CALL (10000)	off
input(1) = IN1 (10001)	on
input(2) = IN2 (10002)	on
input(3) = KEYB11 (10003)	off
input(4) = KEYB22 (10004)	off
input(5) = KEYB33 (10005)	off
input(6) = KEYB44 (10006)	off
input(7) = RESTORE (10007)	off
output(0) = OUT1 (20000)	off [toggle]
output(1) = OUT2 (20001)	off [toggle]

FIGURE 61. STATUS REPORT

As well as displaying the device’s status, it also allows you to force a call [call], perform an audio test [test], activate an emergency situation [activate] and activate the outputs [toggle].

The “**refresh**” option shown in the Figure 61 refreshes the information being displayed.

6.6 Viewing the video

If you have an IPefono with a camera, if it is connected and configured, you can view it by selecting the option “**Video camera**” on the home screen.

Main menu

Setup wizard

Configuration

File manager

System status

Video camera

System Tools

Device pinout

FIGURE 62. VIEWING THE VIDEO

6.7 Customising messages

The system can present the user messages, which are WAVE files recorded with A-law or μ -law PCM encoding. These files are sent to the device by using the “**Upload file**” option shown in [Figure 56](#) or “**File manager**” option on the home screen (see [Figure 6](#)).

With this a new window appears in which the **Select file** button allows you to select the location of the WAVE file that you want to send.

File name	Size (bytes)	Action
config.ini	5218	[delete] [download]
FueraServicio.wav	19526	[delete] [download] [play]
IPefono-HQ.fmw	353370	[delete] [download]
FueraServicioFem.wav	18584	[delete] [download] [play]
LlamadaRecibida.wav	35128	[delete] [download] [play]
LlamadaRecibidaFem.wav	32496	[delete] [download] [play]
Peligro.wav	19790	[delete] [download] [play]
0	4436	[delete] [download] [play]
1	3452	[delete] [download] [play]
2	4168	[delete] [download] [play]
3	3900	[delete] [download] [play]
4	4258	[delete] [download] [play]
5	4792	[delete] [download] [play]
6	4972	[delete] [download] [play]
7	4432	[delete] [download] [play]
8	3586	[delete] [download] [play]
9	3726	[delete] [download] [play]
punto	3514	[delete] [download] [play]

Upload file: Ningún archivo seleccionado

FIGURE 63. “FILE MANAGER” OPTION

The files on the device can:

- ❑ Be deleted with the “**delete option**”
- ❑ Be downloaded with the “**download**” option
- ❑ Be played (in the case of wav files) with the “**play**” option

6.8 Administration and diagnostics tools

During installation and startup, certain problems may arise that prevent the intercom system from working properly. When this happens it is necessary to have tools that allow you to correctly diagnose the cause so that you can fix any issues that are found.

You need a Telnet client to be able to access these tools. You can use any such tools that are installed on your system, or any other commercial or free version, such as PuTTY (<http://www.putty.org>).

To connect to the intercom via the *telnet* client, you need to enter the IPefono's IP address and the port used for Telnet, 23 by default.

You can quickly view all of the available commands and a description of each by using the "**help**" command.

6.8.1 Traces

When the connection has been established, run the "**trace**" command and you will be shown a list of the different types of traces available.

```
CMD>trace
Flag Name  Description
-----
system     System information
ip         IP protocol traffic
cgi        Common Gateway Interface
audio      Audio status
sip        Session Initiation Protocol
io         Inputs and outputs
phone      Phone control
video      Video camera
serial     Serial port data
debug      General purpose debugging
```

The main trace for monitoring communications between VoIP terminals and PBXs is "**trace sip**". This activates the monitoring of the control protocol for standard communication between IP telephony devices.

Normally, if everything is working properly, a point-to-point call is traced as follows:

```
CMD>trace sip
Traces CMD>trace sip
Traces ON: sip

CMD>
[TX]
INVITE sip:4251@192.168.22.252 SIP/2.0
Via: SIP/2.0/UDP 192.168.22.41:5060;branch=z9h2154bKhj240hs8a2s777;rport
From: "Entrada1" <sip:4041@192.168.22.41>;tag=2569624584
To: <sip:4251@192.168.22.252>
CSeq: 0 INVITE
Call-ID: 1413477368-959023025@192.168.22.41
Contact: <sip:4041@192.168.22.41>;expires=60
Allow: INVITE, ACK, OPTIONS, BYE, CANCEL, INFO, NOTIFY, SUBSCRIBE
Content-Type: application/sdp
Content-Length: 206

v=0
o=4041 2917173441 2917173441 IN IP4 192.168.22.41
s=-
c=IN IP4 192.168.22.41
t=0 0
m=audio 5004 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:40

[sip] SipSetStatus(SIP_CALLING)
[RX]
SIP/2.0 100 Trying
```

The aim of this is not to explain the SIP protocol, as that is very complex, but it can be useful for a quick diagnosis, especially if you look at the following points:

1. Traces of messages transmitted (**[TX]**) should appear, otherwise it is likely that the call destination is incorrect or has not been configured.
2. Messages received (**[RX]**) should appear. Otherwise this indicates that the remote terminal is not receiving the packets sent by the IPefono. You will have to review the configuration of the network and the SIP ports.

3. All SIP commands are given one or more response codes, indicating how the call is progressing or that an error has occurred. The code **200** (*correct*) informs you that it has been successfully received, the **1XX** codes (*provisional*) that the command is being processed, **3XX** codes (*redirection*) that the call has been diverted and **4XX** (*client failure*), **5XX** (*server failure*) and **6XX** codes (*global failure*) that an error has occurred.

The following traces are also useful:

- ❑ **phone**: This allows you to check that the behaviour of the intercom is configured correctly.
- ❑ **video**: Shows communication with the connected camera.
- ❑ **io**: Indicates the processes being performed at the outputs and shows any changes to the status of the inputs.

These types of traces are cumulative, so they can be activated and appear together. To cancel current trace monitoring, simply execute the “**trace off**” command.

6.8.2 Network commands

When the IPefono is not communicating properly with the remote device, you can perform a number of tests to see where the issue is.

- ❑ **ping <destination IP address>**: This is used to check whether there is IP connectivity, without taking into account the configuration of the protocols and ports.
- ❑ **arp -a**: This allows you to view the table that lists the IP address with the physical address or MAC.
- ❑ **netstat**: Displays the table with the IPefono’s TCP/IP connection ports and their status.

6.8.3 Audio commands

If there are problems with the audio, you can perform a number of tests by using the following commands.

- ❑ **play <message file>**: Plays a message over the speaker.
- ❑ **audiotest**: Performs an audio test with the speaker and microphone.
- ❑ **call**: Makes a call as if you have pressed the call button.

6.8.4 Video commands

These commands check the connection and that the camera is working.

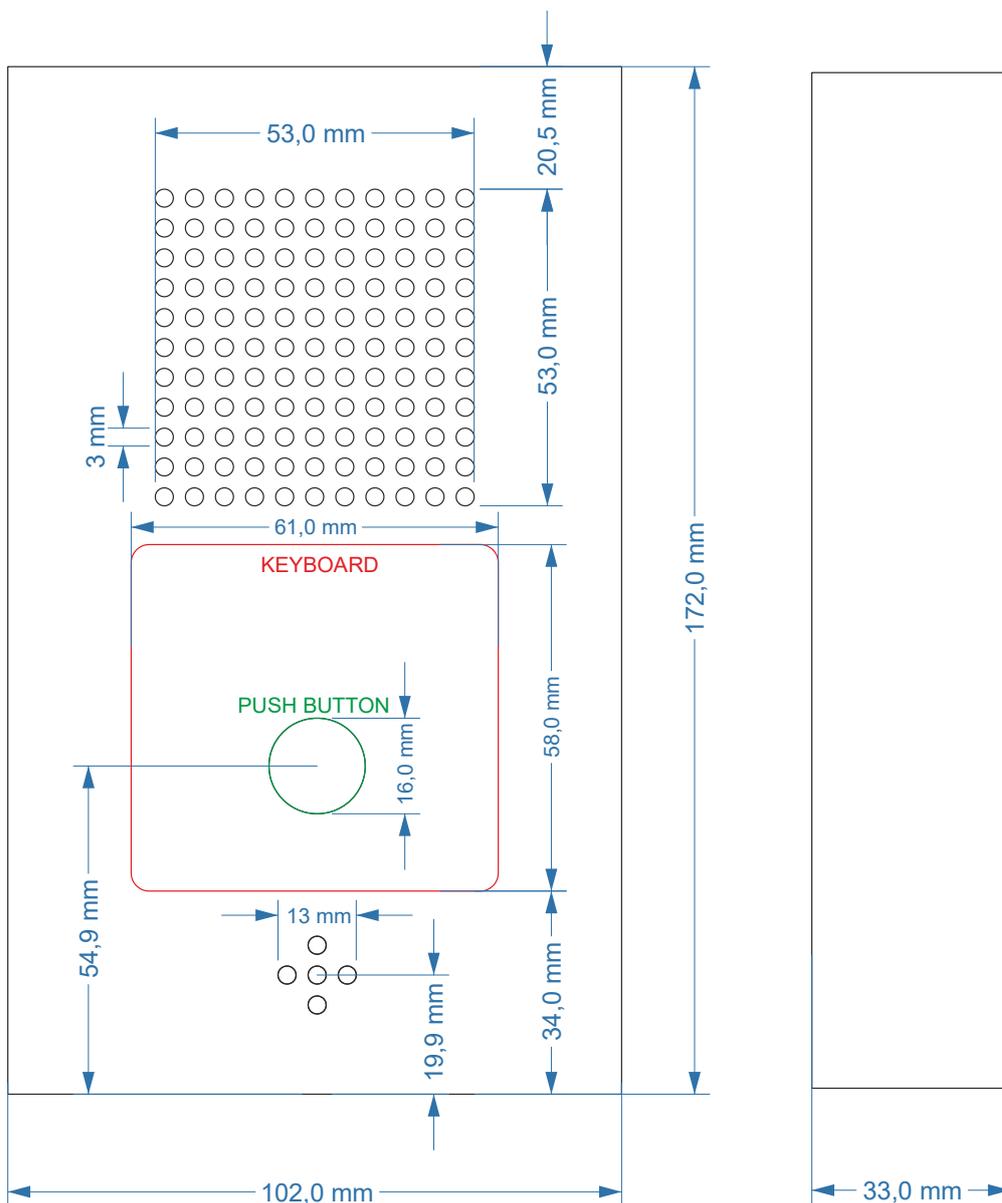
- ❑ **video version**: Prints out the version and serial number of the camera.
- ❑ **video detect**: Detects the communication speed with the camera.
- ❑ **video init**: Restarts the camera.

7

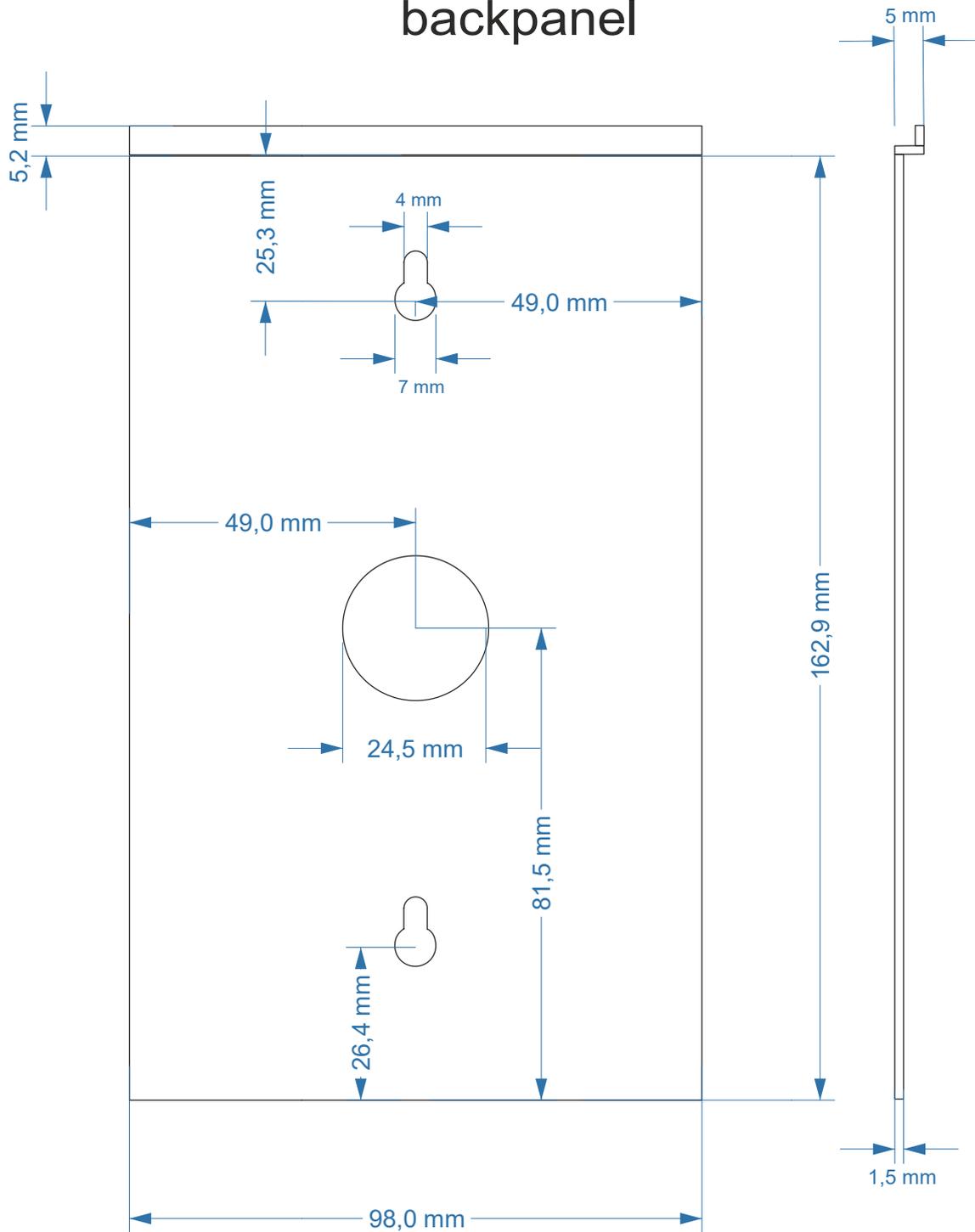
Drawings and dimension

7.1 IPefono Wall Mount

IPefono-WM

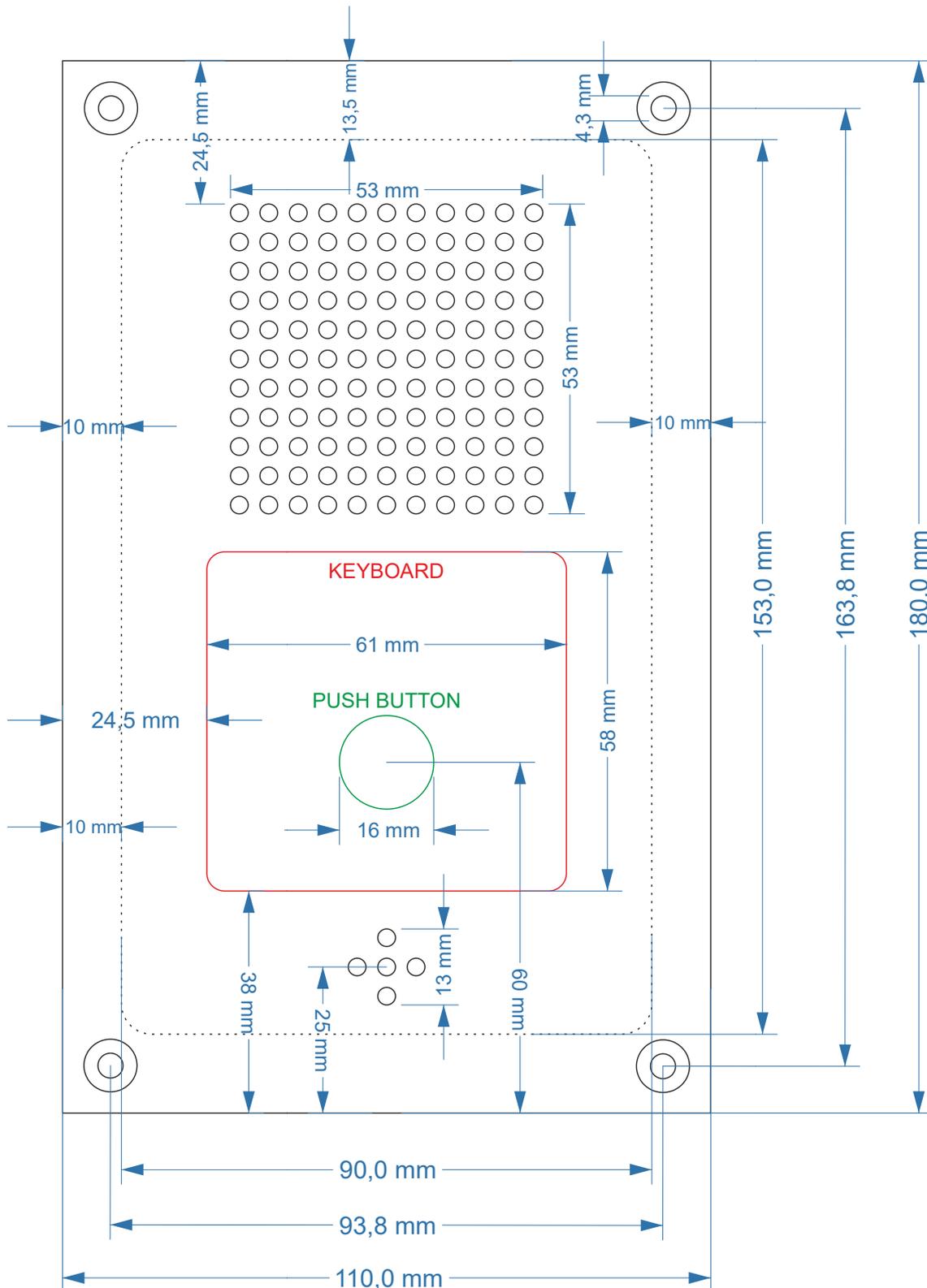


IPefono-WM backpanel



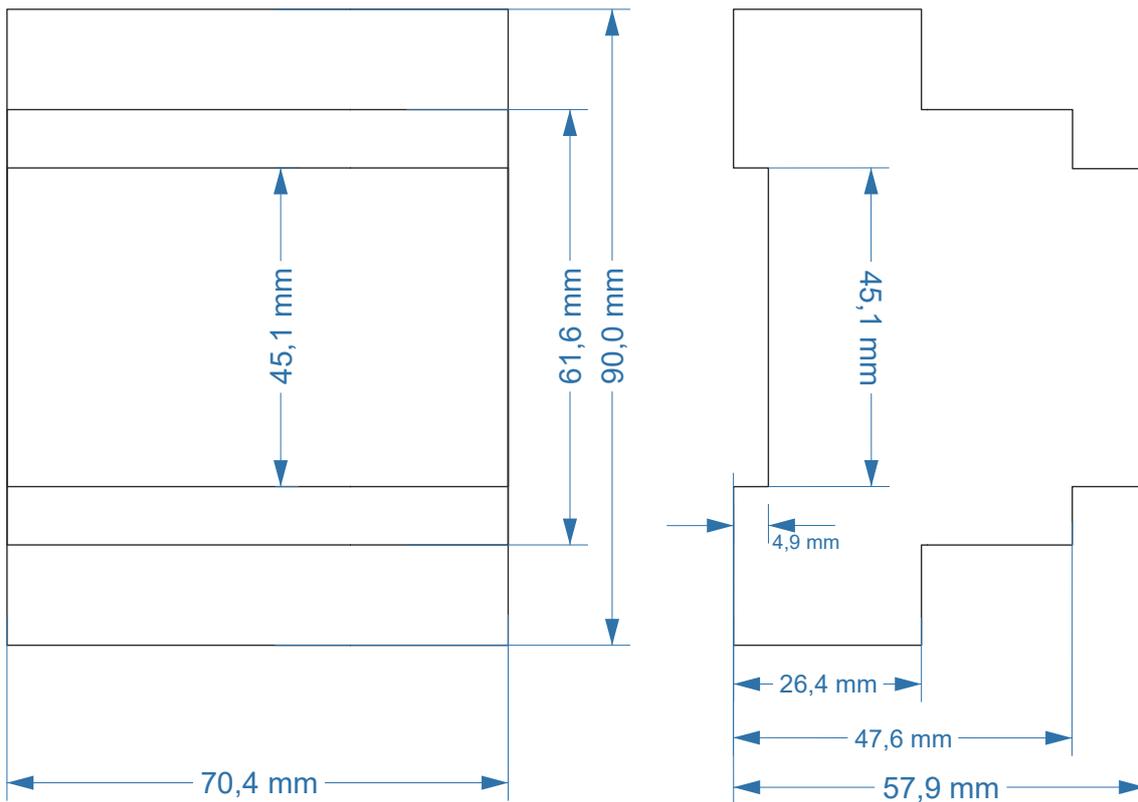
7.2 IPefono Wall Mount Embedded

IPefono-WM embedded



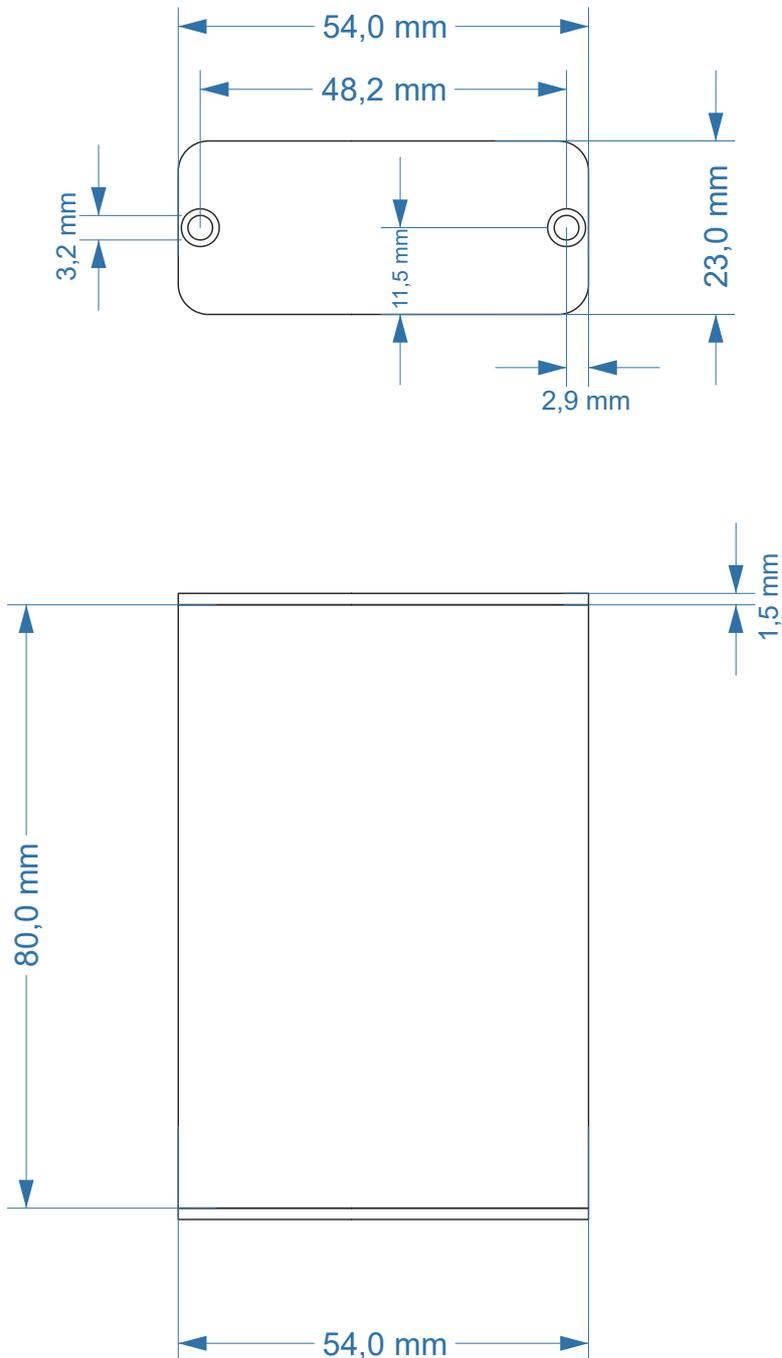
7.3 IPefono HQ with enclosure

IPefono-HQ enclosure



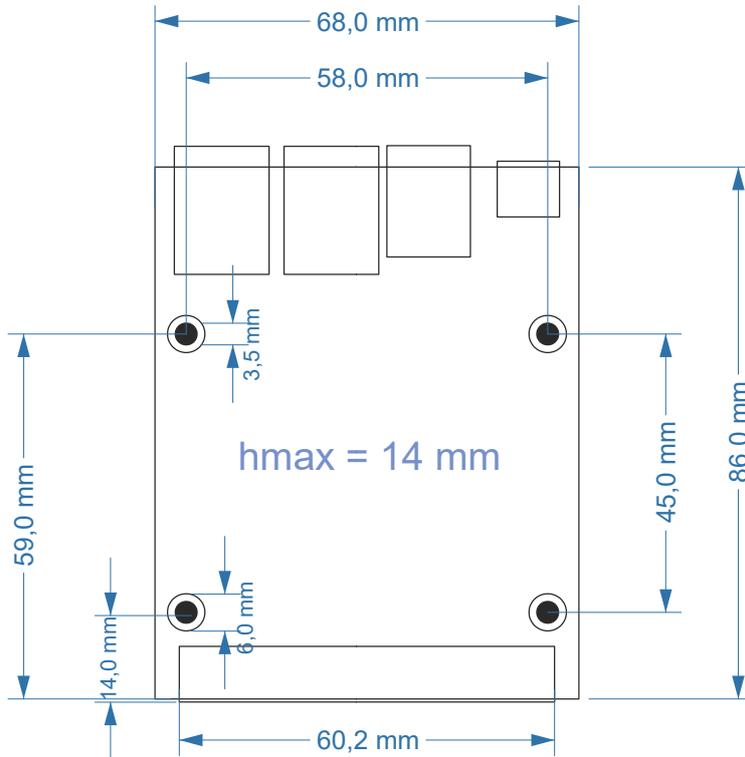
7.4 IPefono LC with enclosure

IPefono-LC enclosure

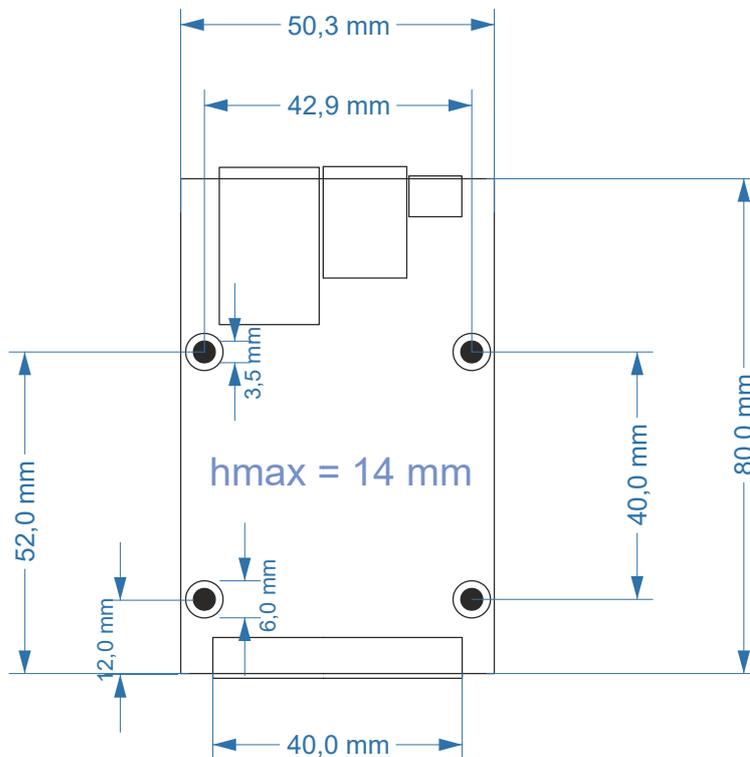


7.5 IPefono HQ and LC OEM

IPefono-HQ OEM

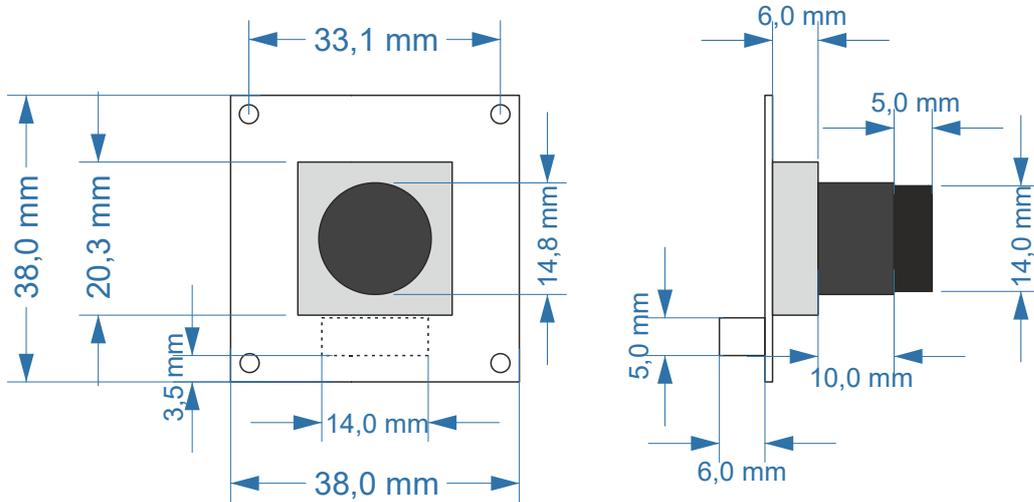


IPefono-LC OEM



7.6 Camera

Camera C429



Camera C339

