

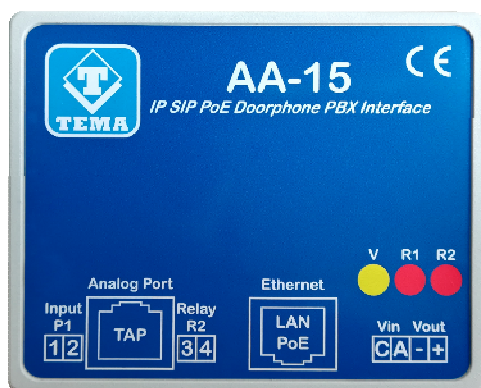


TEMA TELECOMUNICAZIONI S.r.l.
Telecomunicazioni - Elettronica - Microapplicazioni Audiotelefoniche



AA-15SIP

VoIP SIP / PoE Doorphone interface



TECHNICAL – INSTALLATION MANUAL PRELIMINARY DOCUMENTATION

HW Version 1.0 - SW Version 1.0

Recommendations

1. Use only original spare parts and consumables supplied by Tema Telecomunicazioni Srl for this equipment. The company shall not be held responsible for any damage caused by the use of materials that they have not supplied.
2. The device has been carefully manufactured and tested. In any case, the product is not recommended for use in situations in which incorrect operating may result in damage to persons and/or property.
3. We recommend that you carefully read all this manual before starting to use the device.
4. Do not expose the device to sunlight and protect it from sources of heat, dust, humidity and chemical agents.
5. This manual is the property of Tema Telecomunicazioni Srl and any duplication and reproduction, even partial, as well as storage on any type of media is forbidden without written permission from Tema Telecomunicazioni Srl.

Revision	Date	Revision reason	Prepared	Checked/Approved
1	30/11/21	First release	DP	DP, FL



DICHIARAZIONE DI CONFORMITÀ CE

DECLARATION OF CONFORMITY CE

We, **TEMA TELECOMUNICAZIONI SRL** Via C. Girardengo, 1/4 - 20161 MILANO

declare under our sole responsibility that the product:

product name **Modulo Intercom Audio e Video VoIP SIP / PoE**

trade name **TEMA TELECOMUNICAZIONI Srl**

type or model **AA-15SIP**

and accessories

to which this declaration relates is in conformity with the essential requirements and other relevant requirements of the R&TTE Directive (1999/5/EC, 2006/95/EC, 2004/108/EC).

The product is in conformity with the followings standards and/or other normative documents:

HEALT & SAFETY EN 60950-1:2006
 +A11:2009
 +A1:2010
 +A12:2011

EMC EN 55022:2010
 EN 55024:2010
 EN 61000-3-2:2006
 EN 61000-3-3 :2008

MILANO, 30 November 2021

TEMA TELECOMUNICAZIONI SRL
D. Pontillo

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The crossed-out wheeled bin symbol below indicates that this electronic equipment is intended to be disposed in a separate collection and not in an unsorted municipal waste, in order to provide for the treatment of WEEE (Waste Electrical and Electronic Equipment) using best available recovery and recycling techniques. Specific treatment for WEEE is indispensable in order to avoid the dispersion of pollutants and other hazardous substances into the waste stream, while recycling leads to reduction of disposal of waste and the negative impacts on environment and human health. That is, priority is given to reuse of WEEE in its components, subassemblies and consumables. As the final holder, the user has an important role in contributing to reuse, recycling and other forms of recovery of WEEE and is responsible to return this waste in the collection facilities set up by EC Member States and to fulfill other duties in compliance with Directive 2002/96/EC and local laws.

Note: the above information is drawn up in compliance with Directive 2002/96 / EC and Legislative Decree 25/7/2005, n.151, which provide for the mandatory of a separate waste collection system as well as particular methods of treatment and disposal of waste electrical and electronic equipment (WEEE).



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RECOMMENDATIONS

1. It is recommended to read this entire manual before proceeding to the installation of the device.
2. The installation and commissioning of the device can only be performed by specialized technicians.
3. The device is accurately manufactured and tested. In any case, the product is not recommended for use where an error of operations can cause property damage and/or injury to persons.
4. It is expressly not recommended maintenance inside the device which must be carried out by Tema Telecomunicazioni, the removal of the closures will invalidate the warranty and makes accessible internal parts with risk of electric shock.
5. Tema Telecomunicazioni accepts no responsibility for damage to property and/or persons resulting from incorrect use of the equipment or by procedures that do not comply with the instructions in this manual. Tema Telecomunicazioni reserves the right to make modification to the technical and functional specifications at any time and without any notice.
6. Products powered via PoE (Power over Ethernet) may only be connected with cables coming from the internal network company (inside the building), are not allowed connections LAN cables coming from outside the building.
7. Use for this device only and exclusively original spare parts and consumables supplied by Tema Telecomunicazioni. The company is not responsible for damage caused by the use of materials not supplied by the same.
8. Do not expose the unit to direct sunlight, protect from heat, dust, humidity and chemicals.
9. Tema Telecomunicazioni reserves the right to vary the product characteristics for improvement without prior notice. Check the WWW.TEMATLC.IT website for any updates to the latest firmware, manuals, and technical documentation.
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This symbol in the descriptions indicates a general warning or a damage danger to equipment or people.



This symbol in the descriptions indicates an electric shock hazard for damage to the equipment or people.



This symbol in the descriptions indicates useful information or a suggestion for the optimization of the device functionality.

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1. PRESENTATION

The AA-15SIP device allows to interface any traditional doorphone (4 or 5 wires) to an IP-PBX in order to manage the doorphone from any internal extension or, in the absence of an IP-PBX, it can also work with a common SIP phone in P2P (Peer-to-Peer) mode. When a visitor presses the call button of the existing doorphone, AA-15 generate a SIP call to a programmed extension number by putting the visitor in communication with the operator who answered, who has the possibility to operate, with a code from the phone, the relay for opening the entrance gate. A second relay can be activated, from the internal telephone as well, for other services.

When an IP-PBX is present, the calls from the intercom can be routed to an external telephone number, including mobile numbers.

The device has an external contact input that can be used as an alarm dialer: when closed, AA-15SIP calls a programmed number and alerts the event with a customizable voice message.

If there is no IP-PBX, the basic functions (communication and gate opening relay activation) are obtained by connecting a normal IP-SIP phone and programming AA-15SIP in P2P (Peer-to-Peer) mode.

It is possible to connect AA-15SIP both to the central unit of the doorphone system or **directly to the internal station (wall-mounted intercom) of the single user.**

AA-15SIP integrates a **PoE** power supply and can therefore be powered on the same cat5/6 LAN cable if coming from a PoE switch. Alternatively, an input for external 230Vac power supply is available (Optional).

The dimensions 76.5x62xH32.5mm (connectors excluded) are extremely compact and the system can be fixed to the wall with the included bracket or on a DIN bar with a special accessory (Optional).

Main features

- ◆ Can be connected with all 4-5 wires doorphones models, DIN rail mounting
- ◆ 1 power open-door relay and 1 auxiliary relay
- ◆ Up to 2 configurable extension numbers (1 for Day Mode, 1 for Night Mode)
- ◆ Relay: configurable opening and closing contact
- ◆ Activity display LED
- ◆ Easy programming via Web browser
- ◆ Compatible with the most popular IP-PBX brands

Total Management via LAN, integrated Web Server

Programming, configuration, loading and listening of audio files, firmware update, audio volume adjustment, backup, configuration reset, device reboot.

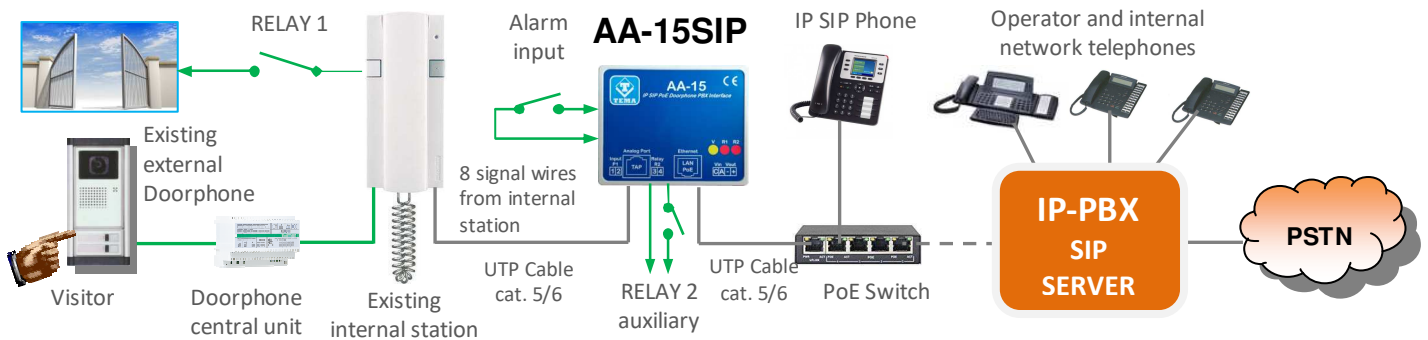
Integration in telephone VoIP SIP and security systems

AA-15SIP integrates perfectly with VoIP telephony systems (IP-PBX) with SIP protocol that can be registered as a normal VoIP phone or in Peer-to-Peer (P2P) mode without a PBX.

Tested with the most popular PBX brands:

SIEMENS/UNIFY - AVAYA - ALCATEL - PANASONIC - SAMSUNG - NEC
LG ERICSSON - WILDIX - AASTRA - ASCOM - NITSUKO - SELTA - PHILIPS
MITEL - YEASTAR - ZYCOO - CISCO - EPYGI ELASTICS - GRANDSTREAM
SHORETEL - NORTEL - SWYX - XORCOM - INNOVAPHONE - NETHESYS
3CX - KALLIOPE - ASTERISK BASED SYSTEMS AND SOFTPHONE APPS

2. TYPICAL OPERATING DIAGRAM



When the visitor presses the button, AA-15SIP generates a SIP call to an internal number, putting the visitor in communication with the operator, who has the possibility to operate the gate opening relay with a code from the phone. In any case, the internal doorphone station remains operational since AA-15SIP works in parallel.

The contacts of Relay 2 can also be brought to the internal station to allow the opening of a second gate or to switch on any lights.

Both relays can be activated not only following the call from the doorphone but also by calling AA-15SIP to the assigned number/IP address, wait for the answer and activate them with codes from the telephone keypad.

In the presence of the IP-PBX, calls from the doorphone can be routed to external phone numbers, including mobile.



IMPORTANT NOTE: the quality of the audio received and transmitted is strictly dependent on the quality of the intercom system, the device does NOT correct any disturbances or low audio quality of the existing intercom system but transfers the audio exactly as it receives it.

3. PACKING LIST

The system consists of the parts included in the following list:

- 1 AA-15SIP interface module
- 1 CD with system documentation
- 1 Quick guide
- 1 ABS bracket for wall mounting + 2 screws
- 1 RJ45 1.5mt LAN cable
- 1 RJ45 cable for TAP socket



NOTE: an optional kit is available for fixing the AA-15S module on a DIN bar, code AA-697/DIN

4. GENERAL FEATURES

- Sending a telephone call upon detection of an intercom call
- Programming via Web interface with password protection
- Day/Night/Interval manual or automatic mode for different call destination numbers
- 2 door opener relays for the possibility of activating a second electric lock
- Setting of the Day/Night/Interval operating mode can be performed by telephone or automatically with time slots, the settings will be retained even in the event of a power failure (requires Internet Time Server access)
- Great versatility coupled with ease of use and programming
- Possibility of software / firmware update via LAN
- Possibility of acquiring 1 external contact to the system and warning service with dedicated message
- Manual "Door opener" function from internal button, to be associated with the available input contact

VoIP IP LAN section

- Integration with the local LAN, LAN 100 BaseT Ethernet port with RJ45 connector
- VoIP connection with SIP protocol both in SIP Proxy Server mode (Registration on IP-PBX) and Peer-to-Peer, possibility of PoE (Power over Ethernet) power supply

5. TECHNICAL FEATURES**Generals**

Insertion terminals for wiring	Possibility of using cables up to 1.5mm ² or AWG16
Number of integrated relays	2
Max relay contact capacity	Up to 1A - 30V
Main unit power supply	12VDC / VAC, 900mA max of absorption
Container material	ABS Novodur®
Mounting type	Wall or DIN rail mounting (optional accessory)
Operating temperature	From -20°C to +50°C
Relative humidity	95% non condensing

VoIP

Power supply via PoE	According to IEEE 802.3af (only for system power supply, not for electric locks)
LAN	LAN 100 BaseT Ethernet Port
Supported VoIP protocols	SIP v2
Supported modes	SIP Server or Peer-to-Peer modes
Protocols	IP, TCP, UDP, HTTP, TELNET, SIP, RTP
Bandwidth	300 – 3400 Hz (7KHz with G722 codec)
Audio codec	G711μ, G711a, G722
Echo suppressor	Yes
Technology	MIPS 560MHz Processor, 128MB Ram, 32MB Flash

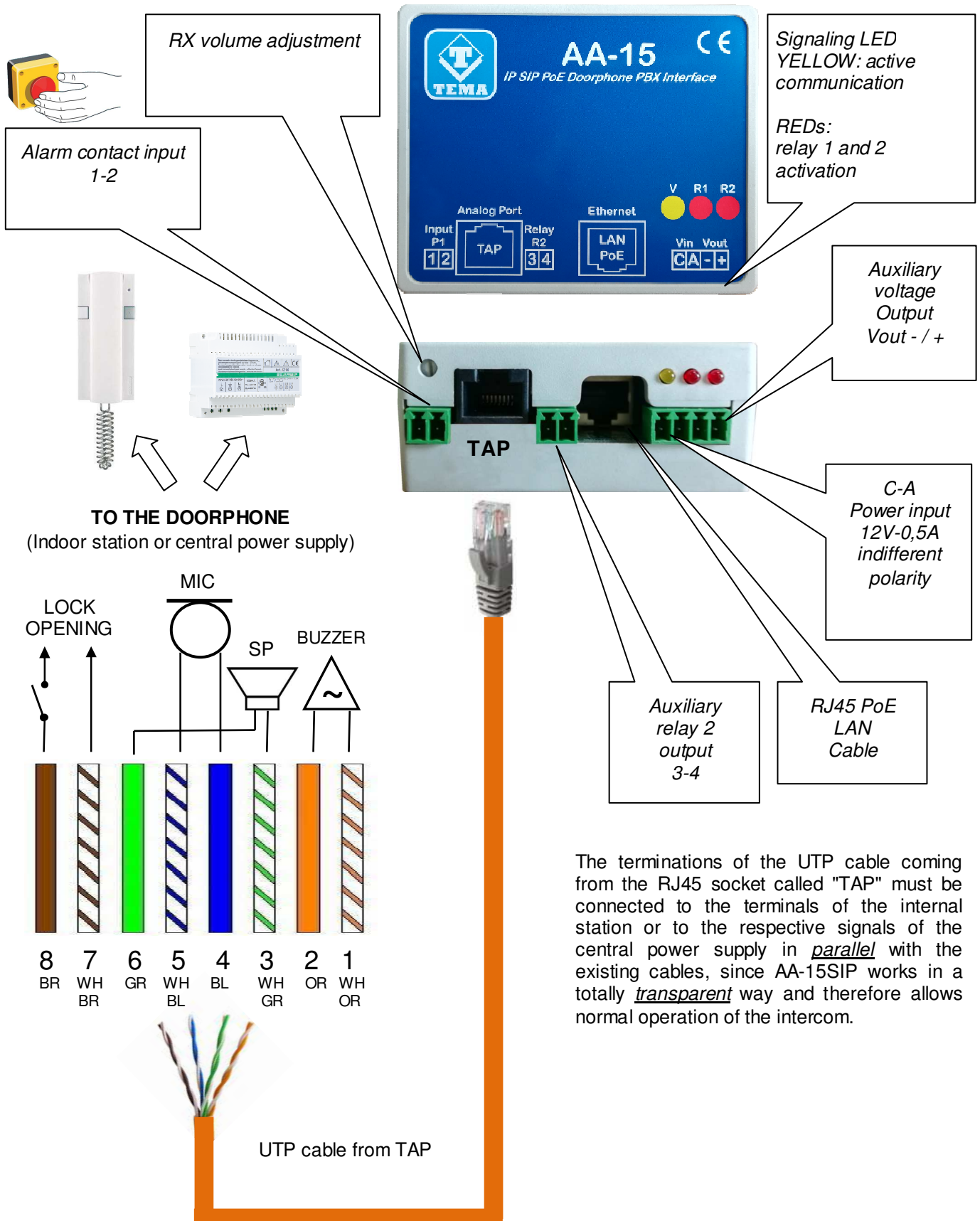
Inputs

Number of acquisition inputs	1 for alarm/call
------------------------------	------------------

6. INSTALLATION

6.1. CONNECTION

The connection of the device must only be carried out by qualified technical personnel.



The terminations of the UTP cable coming from the RJ45 socket called "TAP" must be connected to the terminals of the internal station or to the respective signals of the central power supply in parallel with the existing cables, since AA-15SIP works in a totally transparent way and therefore allows normal operation of the intercom.

Removable screw terminal blocks

1-2 Input P1 Terminal for detecting the alarm contact (1 = contact, 2 = GND)

3-4 Relay R2 Auxiliary relay, range 30V-1Amp, normally open contact

C, A Terminal for powering the system, irrelevant polarity

+ Vout Terminal from which it is possible to draw power, POSITIVE

- GND Terminal from which it is possible to draw power, NEGATIVE



To the terminals 1 and 2 must only be connected to a relay or button contact that is free of any voltage to avoid permanent damage to the device.



At the "+Vout" output there is a + 12Vdc voltage when the device is powered via PoE, otherwise, at this output the voltage will be the same as at the terminals A-C. The use of "+Vout" is permitted as long as it is not exceeded a maximum 200mA current consumption and the load has a protection on the input current. Improper use of this output will permanently damage the unit.

LAN RJ45 port

The module requires a cable for connection to the LAN. If the cable also carries PoE power supply, it will not be necessary to supply the module with other power sources. PoE power supply can coexist with any power supply from an external power supply.



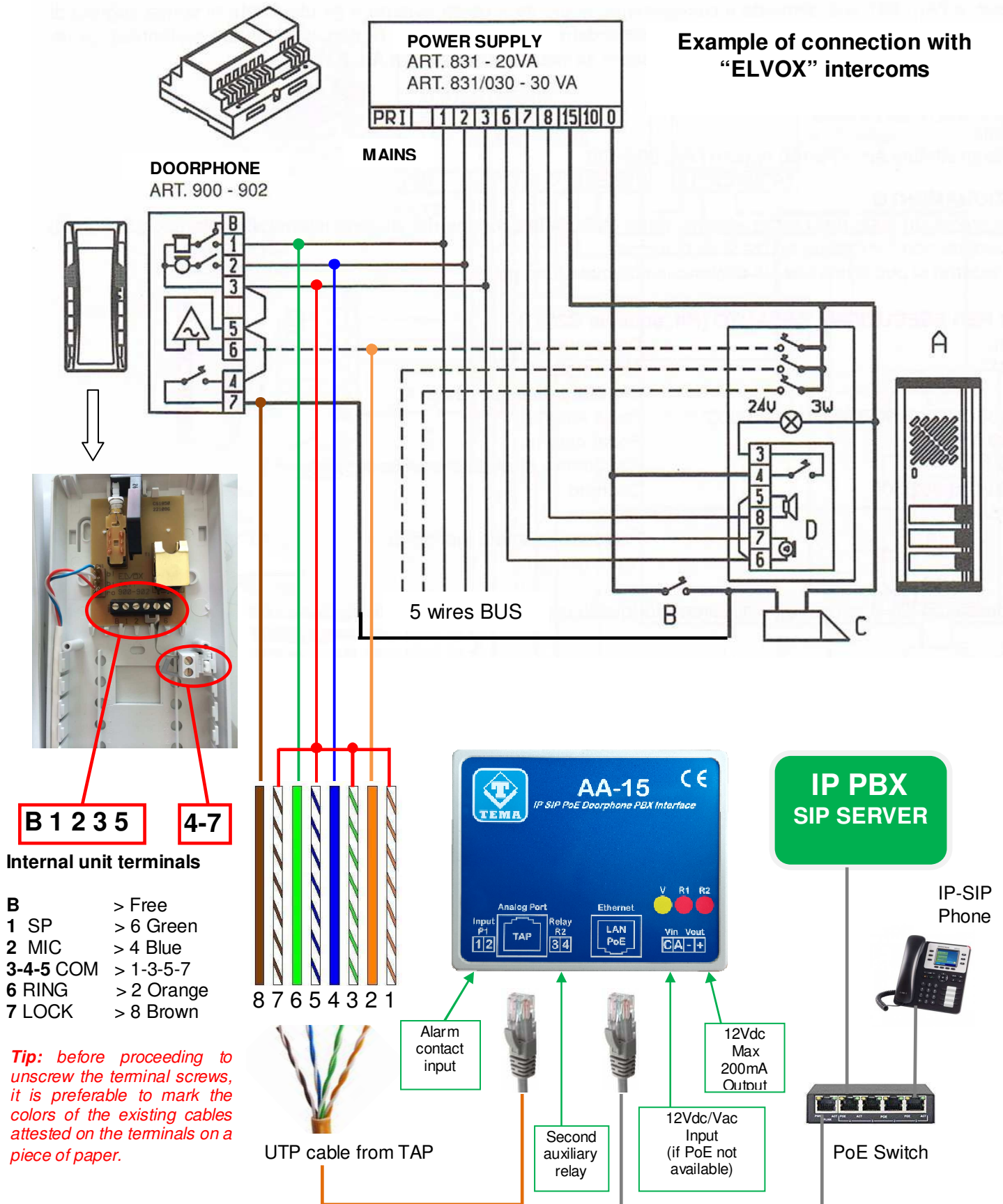
Devices powered via PoE (Power over Ethernet) may only be connected with cables coming from inside the building, they are not allowed connections to LAN cables coming from outside the building.

TAP Analog Port RJ45

All the signals connected to the traditional intercom are brought to this input to be processed by the internal electronics and converted into VoIP-SIP protocols.



IMPORTANT NOTE: the quality of the audio received and transmitted is strictly dependent on the quality of the intercom system, the device does NOT correct any disturbances or low audio quality of the existing intercom system but transfers the audio exactly as it receives it.



AA-15SIP is connected with a UTP cable from the RJ45 "TAP" socket on the terminals of the indoor station (in parallel with the wires already connected to the intercom system). The signals for the AA-15SIP interface are taken from the group of 6 terminals. Terminal 4-7 carries the first relay inside AA-15SIP which activates the lock following a code dialed on the internal telephone keypad. The device works transparently while maintaining the functionality of the indoor station which can continue to be used normally. A second relay is available for other functions. An alarm input is available which, if closed, makes a phone call and sends a message.

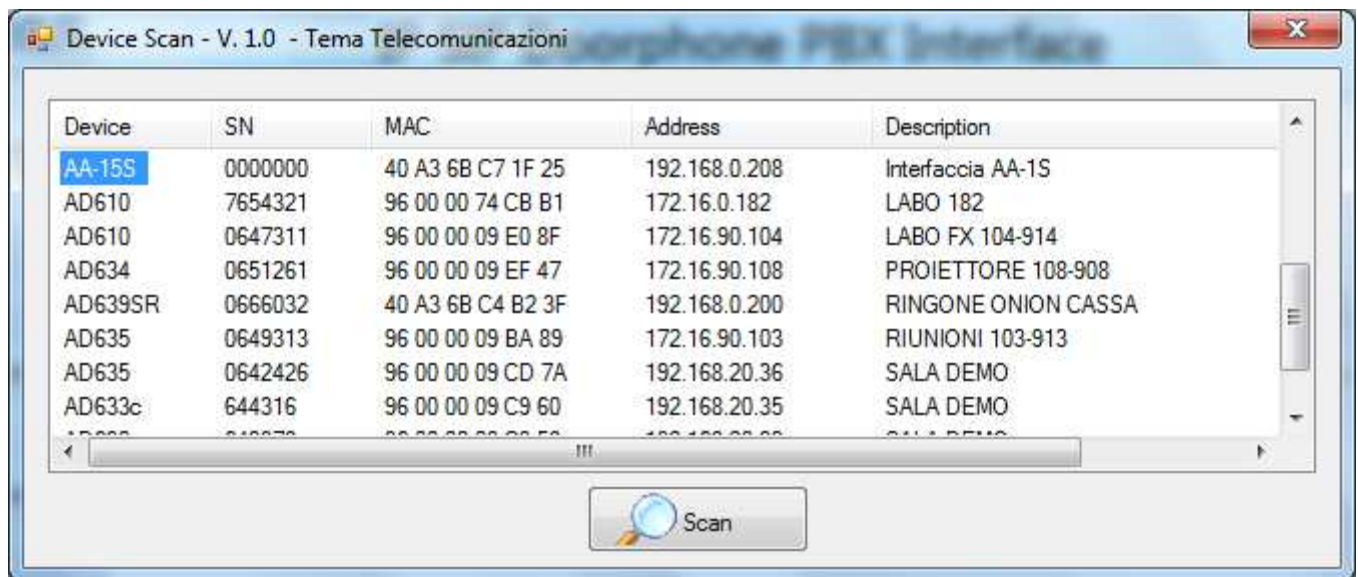
NOTE: see the wiring diagrams of other brands and models of interphones in the appendices at the bottom of the manual.

7. PROGRAMMING

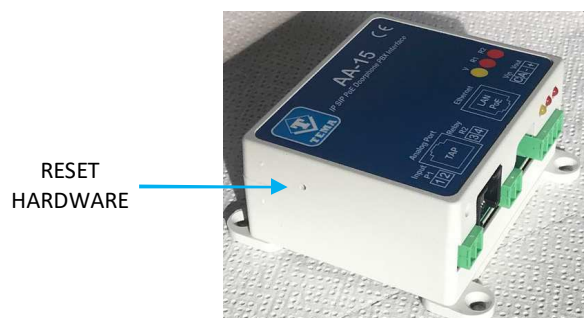
7.1. Preparation for the system parameters programming

Programming is done via the WEB interface. To gain access, simply connect an Ethernet cable from a PC or from a switch to the LAN port of AA-15SIP.

The system is provided in DHCP (dynamic setting of the network address) for which the IP address is automatically assigned by the local DHCP server. To find out the assigned IP address, or, in the absence of a local DHCP server to be able to program one, use the appropriate TEMA network device scan program supplied with it called **"devicescan-tema.exe"**.



This software will show all the TEMA devices present in the LAN with their respective serial numbers, MAC address, IP address, description of the devices. By double clicking on the device name it will be possible to manually change the IP address and Netmask.



Please note that if the network setting is changed from Dynamic to Static, you will need to make sure you are using the same subnet configured on your computer.

If, by mistake, an invalid address has been entered with the result of no longer being able to reach the device, it will be necessary to restart AA-15S 2 times, the second time AA-15SIP notices the invalid address and sets itself to DHCP to receive a dynamic address from the server. At the end you can set a static IP address again.

It is possible to reset the device to factory settings with a **HARDWARE RESET** by using a small screwdriver or a clip in the side hole and holding down the internal button for at least 7 seconds.

7.2. Access to programming

To program the device, it is sufficient to use a standard browser such as Explorer, Firefox, Chrome or others. The user/password with which to connect are **master/master**. logged in, it is possible to change the administrative password for the maximum safety of your device, see par. 7.9.

The use is very simple and intuitive, the menu for selecting the functions to be programmed is always visible on the left, while the configuration mask active at that time is shown on the right. Each change will be confirmed with the **"Apply"** or **"Save"** buttons. Closing the browser or changing pages without selecting these buttons **will make any changes lost**.

As soon as the device is accessed, a mask will be presented which summarizes the general status of the system:

The screenshot shows the web interface for the TEMA AA-15SIP device. The interface has a light blue header with the TEMA logo and the title 'IP SIP Doorphone PBX Interface'. A 'Logout' button is in the top right. On the left is a vertical menu with options: Home, Network, SIP settings, General Settings, Day/night mode, Alarm, Outputs, Log, Maintenance, and Manual. The main content area is divided into sections: 'GENERAL INFORMATION' (Serial number: 0000000, MAC address: 40 A3 6B C7 1F 25, AD version: 1.1.0, Mode: Day), 'LAN' (IP address: 192.168.0.11, Subnet mask: 255.255.255.0, Gateway: 192.168.0.1), and 'SIP' (Account: REGISTERED, highlighted in green). On the right side, there is a 'TIME: 14:44' display and a small image of the physical device showing its ports and status LEDs.

Any changes do not require restarting the device (except changing the IP address and updating the software).

7.3. Network parameters

On this page it is possible to set the network parameters, such as IP address, netmask, etc. :

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IP SIP Doorphone PBX Interface

Logout

AA-15SIP
Home
Network
SIP settings
General Settings
Day/night mode
Alarm
Outputs
Log
Maintenance
Manual

LAN

Connection type : ☐ Dynamic ☒ Static
IP address : 192.168.0.11
Subnet mask : 255.255.255.0
Gateway : 192.168.0.1
Primary DNS : 8.8.8.8
Secondary DNS : 8.8.4.4
Time server : ntp1.inrim.it
Test address (ping) :

DISPOSITIVO

Description : Interfaccia AA-1S

Save

TIME: 14:45

In case of static network configuration, gateway and DNS are only necessary if you want the device to be able to access the Internet (for example to get the current date/time, in the example from the *ntp1.inrim.it* website).

The changes to the network settings are taken over by the device until the next reboot. Once you have completed the configuration steps so be sure to reboot the system and if necessary, change the network segment of your computer so that it can connect to the new address.

- Connection type: indicate the type of connection, Dynamic (DHCP) or Static.
- Subnet mask: indicate the subnet mask to be assigned to AA-15SIP.
- Default gateway: indicate the gateway that can enable AA-15SIP to access to internet.
- Primary and secondary DNS server: indicate the addresses of two DNS servers you want to use to resolve IP addresses.
- Time server: indicate the address of the server you want to use for time synchronization device.
- Test address (ping): if inserted, this address is used by the system to verify the correct operation of the network connection.
- Device description: text that will appear on the search mask of the supervision Tema Adam software (for this device it is not necessary).

7.4. SIP parameters

Once the network parameters have been set correctly, you must configure the connection with the VoIP SIP PBX:

- SIP server address: specify the IP address of the SIP PBX to which AA-15 should connect.
- Port: is the port number of the SIP PBX with which AA-15 must operate. Generally it is the 5060, but some stations use a different port.
- Domain: enter the domain to which register the system.
- Outbound proxy: some PBXs require that the extension number to call is followed by the proxy address. In most cases this field can be left empty.
- Local SIP port: generally the 5060 port is used, but it is possible to specify a different number (i.e. for particular routing tables).
- Expiration: indicates how many seconds the device should check for correct registration of the extensions.
- User/password: credentials for the registration of the extension (which obviously must have been already created on the PBX).

7.5. General parameters

In this section it is possible to set parameters for general use, such as call duration, audio volume, and so on.



IP SIP Doorphone PBX Interface


Logout

AA-15SIP

Home

Network

SIP settings

General Settings

Day/night mode

Alarm

Outputs

Log

Maintenance

Manual

CALL PARAMETERS

Communication time :	<input type="text" value="60"/>		
Call attempt time :	<input type="text" value="15"/>		
Answer time :	<input type="text" value="0"/>		
DAY number :	<input type="text" value="779"/>	NIGHT :	<input type="text" value="779"/>

AUDIO

Communication volume :	<input type="text" value="9"/>
Microphone volume :	<input type="text" value="9"/>
Echo limiter :	<input type="text" value="0.01"/>

Save

TIME: 14:48



- Communication time: once in connection with the extension, the call is still cut down after the set time.
- Call attempt time: duration in seconds of the internal call attempt.
- Answer time: when the interface is called, it responds after the set time. If it is 0, it answers immediately, if 999 it never answers.
- DAY / NIGHT number: extension number to call when the system is in DAY or NIGHT mode. The mode can be set manually or automatically (see below). If the system is not registered in a PBX, it is also possible to enter the IP address of a SIP device to contact (Peer-to-Peer call).
- Communication volume: sets the volume of the audio played to the external user. Values range from 0 (very low) to 9 (very high).
- Microphone volume: sets the sensitivity of the microphone and consequently the volume of the audio reproduced towards the user within the company. Values range from 0 (very low) to 9 (very high).
- Echo Limiter: this parameter indicates the audio level beyond which the voice exchange between the caller and the callee can be carried out. Leave the default value.

7.6. Set of the Day/Night operating mode

In the operating mode, it is possible to set the mode to DAY/NIGHT, either manually or automatically following time slots per week:

IP SIP Doorphone PBX Interface

AA-15SIP

MODE

Current mode : ☒ Automatic ☐ Day ☐ Night

AUTOMATIC mode code : *0

DAY mode code : *3

NIGHT mode code : *4

NIGHT BANDS

Day	from	to	from	to	from	to	from	to
Monday	23:59	08:30						
Tuesday								
Wednesday	14:00	16:00						
Thursday								
Friday								
Saturday								
Sunday	12:44							23:55

Save

TIME: 14:49

Logout

AA-15SIP

- Current mode: allows to set the current mode of operation.
- Change mode codes: enter the codes to change mode remotely (from any internal phone, call the Speaker and, at the answer, via DTMF type the code corresponding to the selected mode). The codes must consist of 2 characters.
- Night bands: in the case of automatic operation, it is possible to enter up to 4 bands per day. If the current time is in any band, the system sets itself to NIGHT mode. Outside of any time band, it is in DAY mode.

7.7. Relay and alarm settings

In this page it is possible to program the input from an external contact:

IP SIP Doorphone PBX Interface

AA-15SIP

ALARM INPUT CONFIGURATION

Function active : ☒

DAY number : 447 NIGHT : 447

Attempts : 1

Separation : 30

Delay : 0

Alarm stop code : 43

Input inversion : ☐

Save

Logout

TIME: 14:50

AA-15SIP

- **Function active:** indicates whether to monitor the status of the alarm input or not
- **DAY/NIGHT number:** number or IP address of the telephone to be contacted in day or night mode when the alarm condition occurs.
- **Attempts:** number of call attempts (max 999)
- **Separation:** separation, in seconds, between one call attempt and the next (max 999).
- **Delay:** it is the minimum time, in seconds, before validating the input. It could be useful, for example, to mask an open door switch. Supposing the maximum time of a pedestrian gate is 30 seconds, at the end of this time (if the switch would be still active) an alarm will be send to the reception.
- **Alarm stop code:** code to be dialed on the phone called to accept and silence the alarm (max 3 digits).
- **Input inversion:** the contact is normally open and the alarm occurs at the closing of the same. Enabling this flag it is possible to instruct the device to behave in the opposite way (the contact is normally closed and the alarm occurs at the opening).

The system can generate a VoIP call to alert, after triggering the alarm input contact (from auxiliary devices or a key made available nearby), to an operator in charge.

When answered, a pre-recorded message is played back to the called party. It is possible to reverse this logic, so the triggering of the signal can occur following the opening of the contact connected to the input.

AA-15SIP continuously monitors the status of the contact and in case of activation it will start calling the person who will have to manage the detected situation at the programmed telephone number and reproduce the message associated with the event. It is possible to define an acquisition / silencing code of the alarm signal that the called party must enter to inform the device that the alarm signal has been accepted. If the called number is busy or does not answer or in any other case in which AA-15SIP does not receive the acquisition / silencing code, at the end of each single call attempt it will go back to idle and prepare for a new notification of the alarm.

When AA-15SIP receives the silencing code, the reporting of the alarm condition will cease and no further notification calls will be made. In order for the notification phase of an activation to be triggered again, the condition that triggered the previous notification must return to rest. The cycle will restart only when a new contact activation condition occurs. In practice: if a connected contact closes and its activation is detected, the AA-15S starts making alert calls.

The text of the alarm message can be changed from the "Maintenance" web page.

7.8. Outputs

The parameters relating to the activation of the two relay outputs are programmed in this screen:

IP SIP Doorphone PBX Interface

AA-15SIP

OUTPUT 1

Mode: Electric lock

Activation code: #1

Activation time: 2

End call: ☐

OUTPUT 2

Mode: Electric lock

Activation code: #2

Activation time: 3

End call: ☐

Save

TIME: 14:50

AA-15

- **Mode:** indicates the activation mode of the relative relay output. It is possible to choose between "Electric lock" (the relay is activated remotely with a code), "During call" (the relay remains active for the entire duration of the call), "Button press" (the relay is activated when the gate call button is pressed) or "Alarm activation" (the relay is activated following the activation of the alarm).
- **Activation code:** code to activate the relay remotely with the telephone (only valid for "Electric lock" mode).
- **Activation time:** duration of relay activation in seconds.
- **End call:** when the relay is activated remotely, the call is immediately cut down.

The normal default mode is "Electric lock": the visitor intercom at the entrance, AA-15SIP forwards the call to the internal extension and whoever answers the call can decide whether or not to open the gate with the command "#1" (default).

7.9. Maintenance

On this page it is possible to check the current firmware version, update software, and backup / restore the system configuration:



Messages

It is possible to listen to or change the system message associated with the alarm. The "alarm" message is the one reproduced by telephone during the activation of an alarm: when the called party answers, the message is played several times until the silencing code is entered or a timeout expires. "SIP Alert Call Button" function.

Configuration

You can back up the current configuration or restore a previously saved one. It is also possible to reset the configuration to the factory value.

Password Master user

By default, the user name and password for access to the configuration are master/master. The user cannot be changed while his password can. In order to change it, the old password and the new password with confirmation are required.



Make a note of the new credentials introduced in order to be able to log back into the system later!

Firmware

Any new firmware or accessory libraries can be loaded through this section. Any updates are released by Tema's technical laboratory to correct any malfunctions or to extend the services. To update, simply select the file to upload, upload it to the device and restart it. The restart key can be used even if you are not updating.

7.10. Diagnostic log

To identify configuration problems, it is possible to activate a diagnostic summary on the activity of the VoIP channel of the device.

The screenshot displays the web interface for the TEMA AA-15SIP VoIP Doorphone PBX. The interface includes a top navigation bar with the TEMA logo, the title "IP SIP Doorphone PBX Interface", and a "Logout" button. On the left, a sidebar menu lists various configuration options: Home, Network, SIP settings, General Settings, Day/night mode, Alarm, Outputs, Log (highlighted), Maintenance, and Manual. The main content area is titled "ACTIVITY LOG" and contains a list of system events with timestamps and details. Above the log, there are "Clear" and "Start dump" buttons. To the right of the log, the current time is displayed as "TIME: 14:52". Below the time, there is a small image of the AA-15SIP device showing its physical ports and status indicators.

AA-15SIP

ACTIVITY LOG [Clear] [Start dump] TIME: 14:52

```

02/11/21 15:48:24 RNG : status:Registration on sip:172.16.0.88:5060
successful.
02/11/21 16:02:25 RNG : status:Registration on sip:172.16.0.88:5060
successful.
02/11/21 16:16:26 RNG : status:Registration on sip:172.16.0.88:5060
successful.
02/11/21 16:30:27 RNG : status:Registration on sip:172.16.0.88:5060
successful.
02/11/21 10:20:11 WD : ----- WD v. 1.00.0
02/11/21 10:20:11 WD : main thread running
02/11/21 16:43:45 RNG : ----- RN v. 1.0.1
02/11/21 16:43:45 [-] M: ----- AD v. 1.0.3
02/11/21 16:43:45 [-] M: play open 8000 default:CARD=Device
02/11/21 16:43:46 [-] M: 0000000
02/11/21 16:43:46 [-] M: setvol 0 (0-37)
02/11/21 16:43:47 [-] L: status:Ready
02/11/21 16:43:47 [-] L: "445"<sip:445@172.16.0.88> sip:172.16.0.88:5060 445
02/11/21 16:43:47 [-] L: lpc_cmd_register
02/11/21 16:43:47 [-] L: "172.16.0.88"
02/11/21 16:43:47 [-] L: sip:AA-15S@192.168.0.208
02/11/21 16:43:47 [-] C: s_idle
02/11/21 16:43:47 [-] M: led OFF
02/11/21 16:43:47 [2] O: s_attivazione
02/11/21 16:43:47 [0] I: s_idle
02/11/21 16:43:47 [1] I: s_idle
02/11/21 16:43:47 [2] I: s_idle
02/11/21 16:43:47 [0] O: s_idle
02/11/21 16:43:47 [1] O: s_idle
02/11/21 16:43:47 [2] O: s_idle
02/11/21 16:43:47 [-] M: open multicast 239.255.15.1 8001
02/11/21 16:43:47 [-] M: open multicast 239.255.15.2 8001
02/11/21 16:43:47 [-] M: multicast thread running
  
```

The system keeps an activity history, which is deleted when it exceeds about 1 MB in size. To facilitate diagnostics, it is also possible to delete the entire log to start from scratch with the "Clean" button.

8. APPENDIXES

8.1. Use of the AUDACITY software for audio files recording

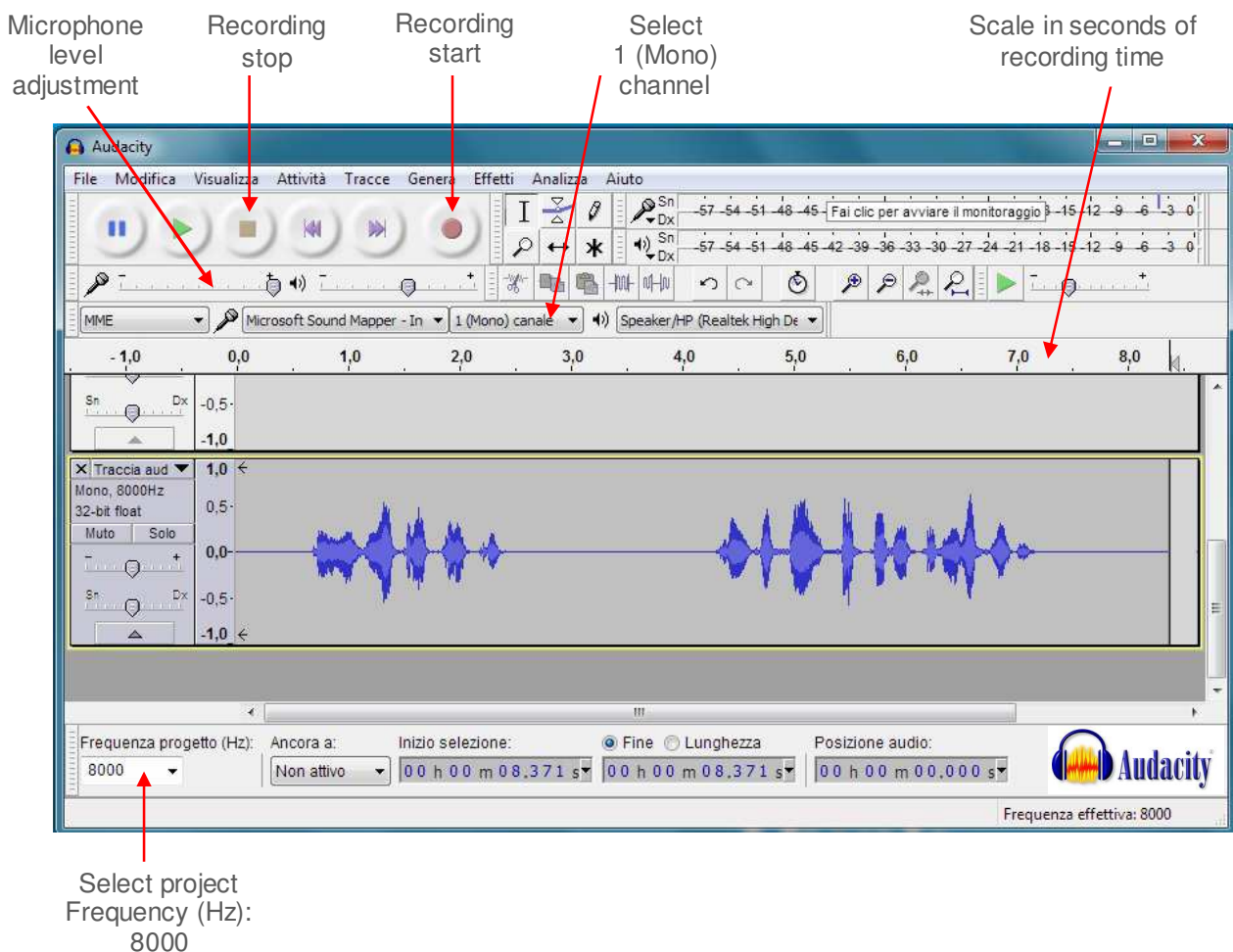
For recording your audio files, it is possible to use one of the free softwares such as AUDACITY downloadable for free from the link <http://www.audacityteam.org/> remembering to record and save audio files in .WAV format at 8KHz-16Bit Mono, other audio file formats will not work with the AD600 series devices. Below there are some informations about basic operations. For more information on the AUDACITY program, please refer to the manufacturer program guide.

Proceed as follows to record a message from the built-in microphone or connected to the PC.

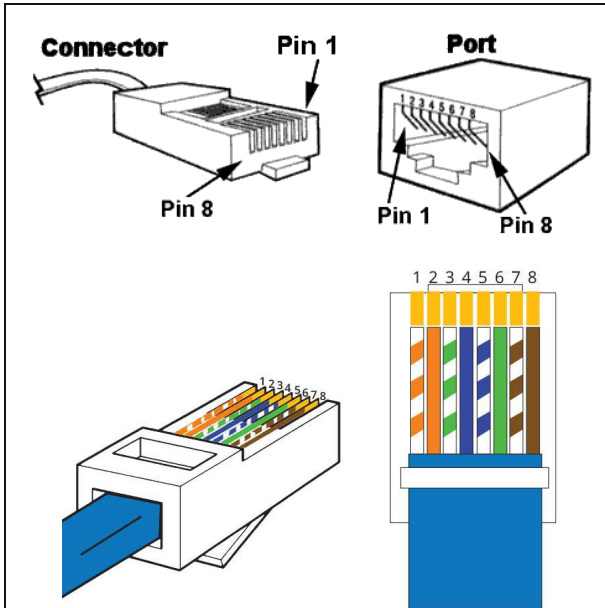
1. Select the 8,000Hz sampling rate, click Start Recording and start talking.
2. To end the recording, click on Stop recording, listen to the recorded audio, and then repeat recording if not optimal. To delete the previous recording, simply select the audio track (crawl from start to finish), press the "Delete" button and repeat the recording. With this procedure it is also possible to delete small parts of silence or sound inside the track.
3. To save the recording click on "File", "Export Audio", give a name and select in the "WAV (Microsoft) 16-bit PCM format" drop-down list. The saved file is ready to be used on all devices in the AD600 series or used with the VLC program (see previous appendix) to be sent as Multicast audio streaming.

The AUDACITY software allows a large number of audio signal processing such as: amplification/reduction of audio levels, duplication of track parts, elimination of silences, insertion of DTMF tones or tones, noise reduction, etc. For optimal use please refer to the product manual.

NOTE: TEMA provides you with its own internal recording studio for the creation of professional prompts with multilingual speakers from texts on customer specifications.

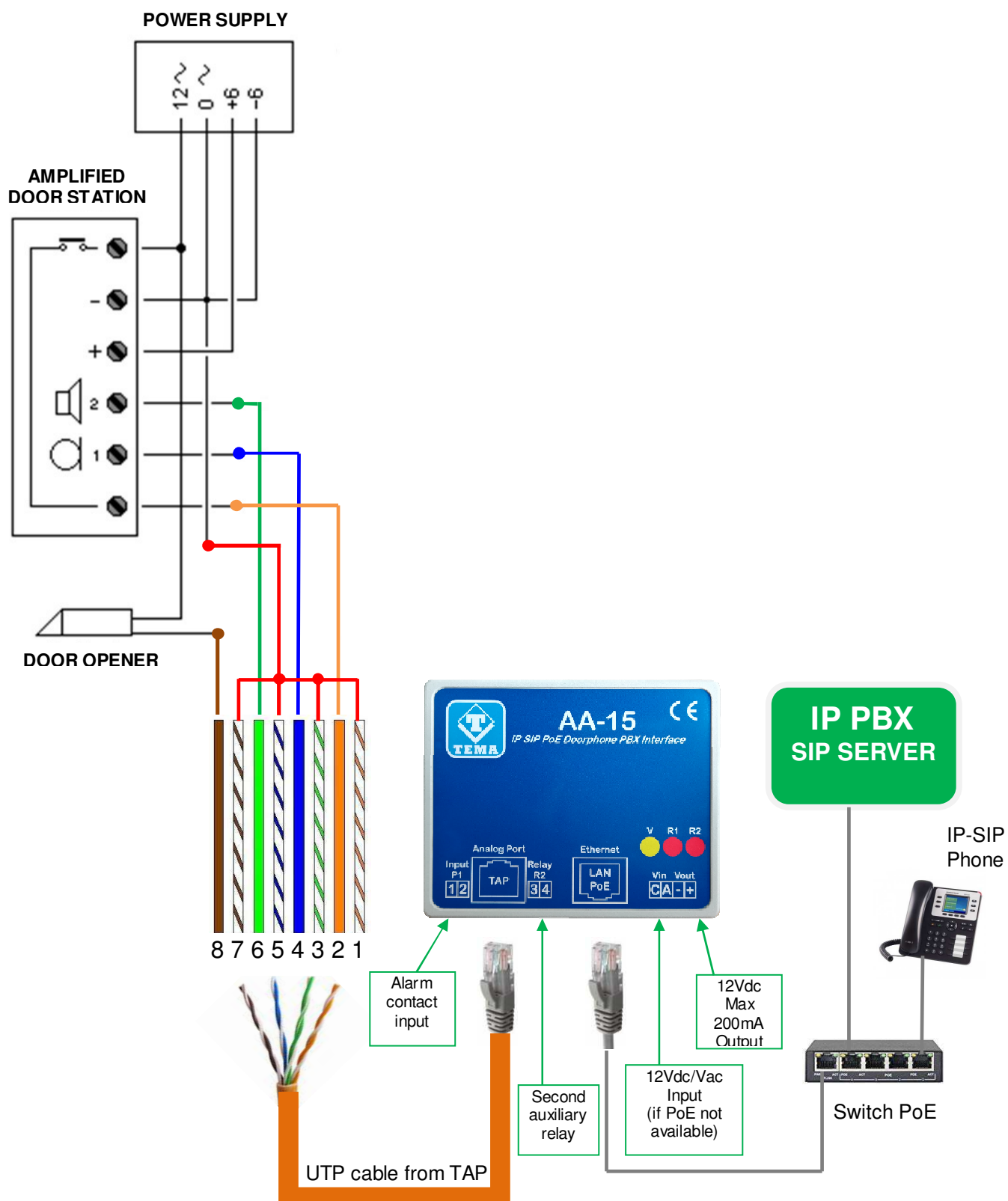


8.2. Cabling of a UTP RJ45 network cable according to the standard EIA568B

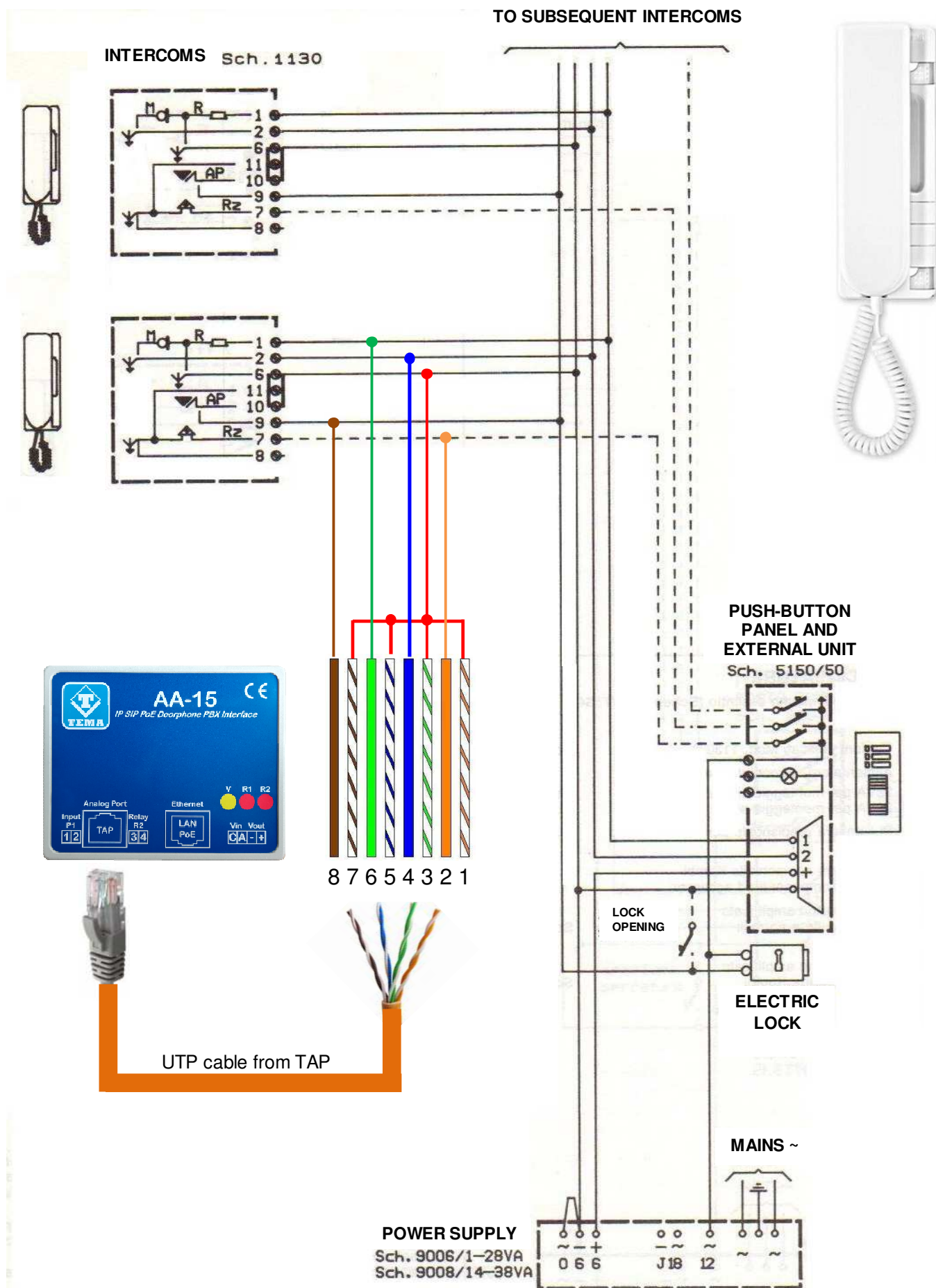
	Pin	Signal	Connector 1	Connector 2
	1	TX+	White/Orange	White/Orange
	2	TX-	Orange	Orange
	3	RX+	White/Green	Bianco/Green
	4	PoE-	Blue	Blue
	5	PoE-	White/Blue	White/Blue
	6	RX-	Green	Green
	7	PoE+	White/Brown	White/Brown
	8	PoE+	Brown	Brown
NB: if the cable colors are different, the right matches must be maintained.				

- Cut the sheath (about one centimeter) to discover the wires.
- To facilitate the process it is possible, by exploiting the elasticity of the sheath, to pull the wires some additional millimeter. (Hold with one hand the wires and with the other pull the smoothing sheath).
- Straighten the wires previously individually pair twisted.
- Compose the color sequence following the pattern.
- Level the length of the cables into place.
- Insert the tightened wires into RJ45 connector holding them between your fingers until they are channeled into the guides inside the connector itself.
- Push well until the wires will touch the bottom of the connector (check in transparency that all the wires are in place).
- Make sure that the sheath has penetrated into the connector for at least 8 mm so that it can also be crimped.
- Place the connector in the crimping tool and tighten all the way. Should be audible a click caused by the outer stop.
- Repeat exactly all the above steps to crimp the cable on the other side.

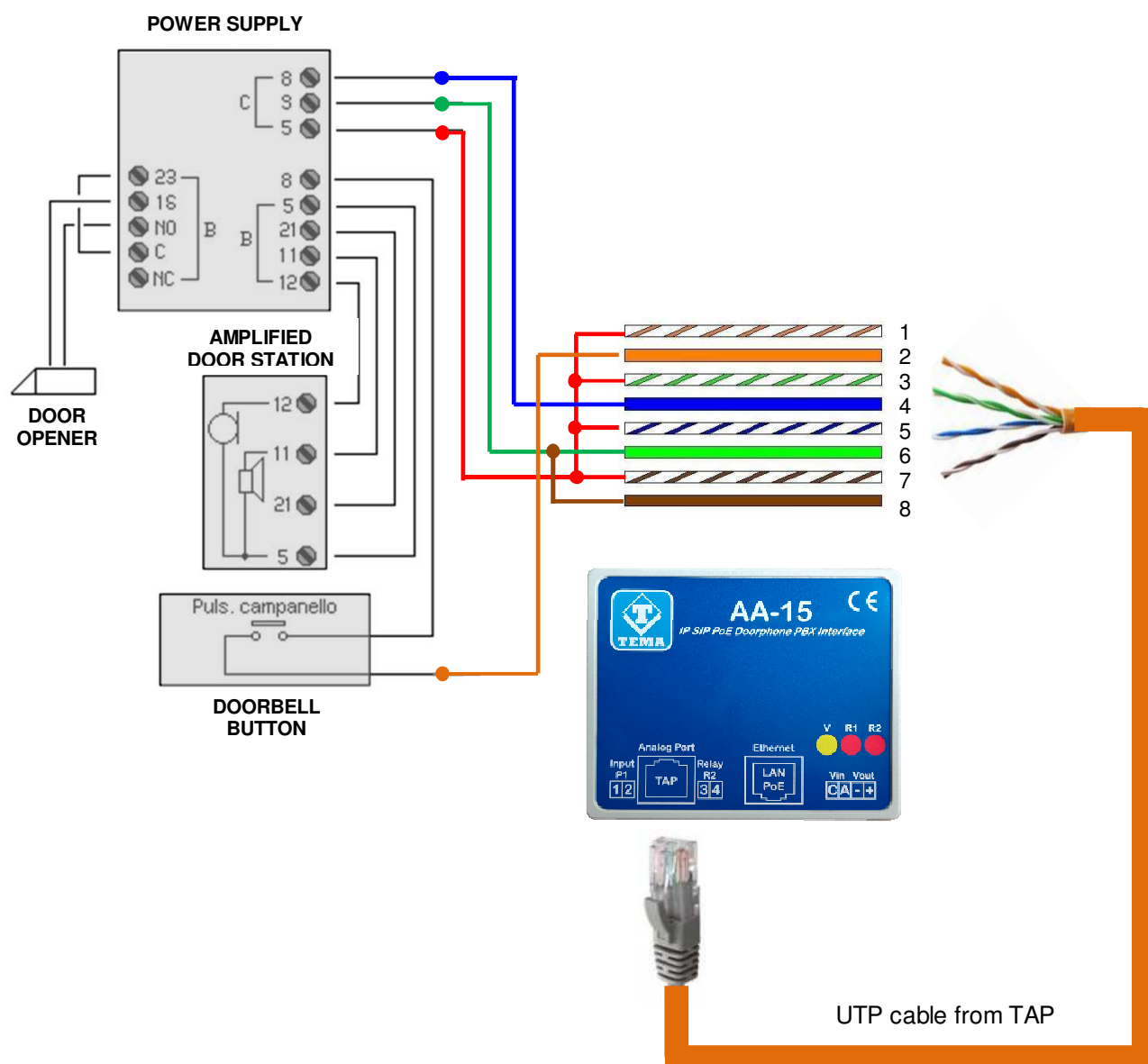
8.3. Connection with URMET intercom



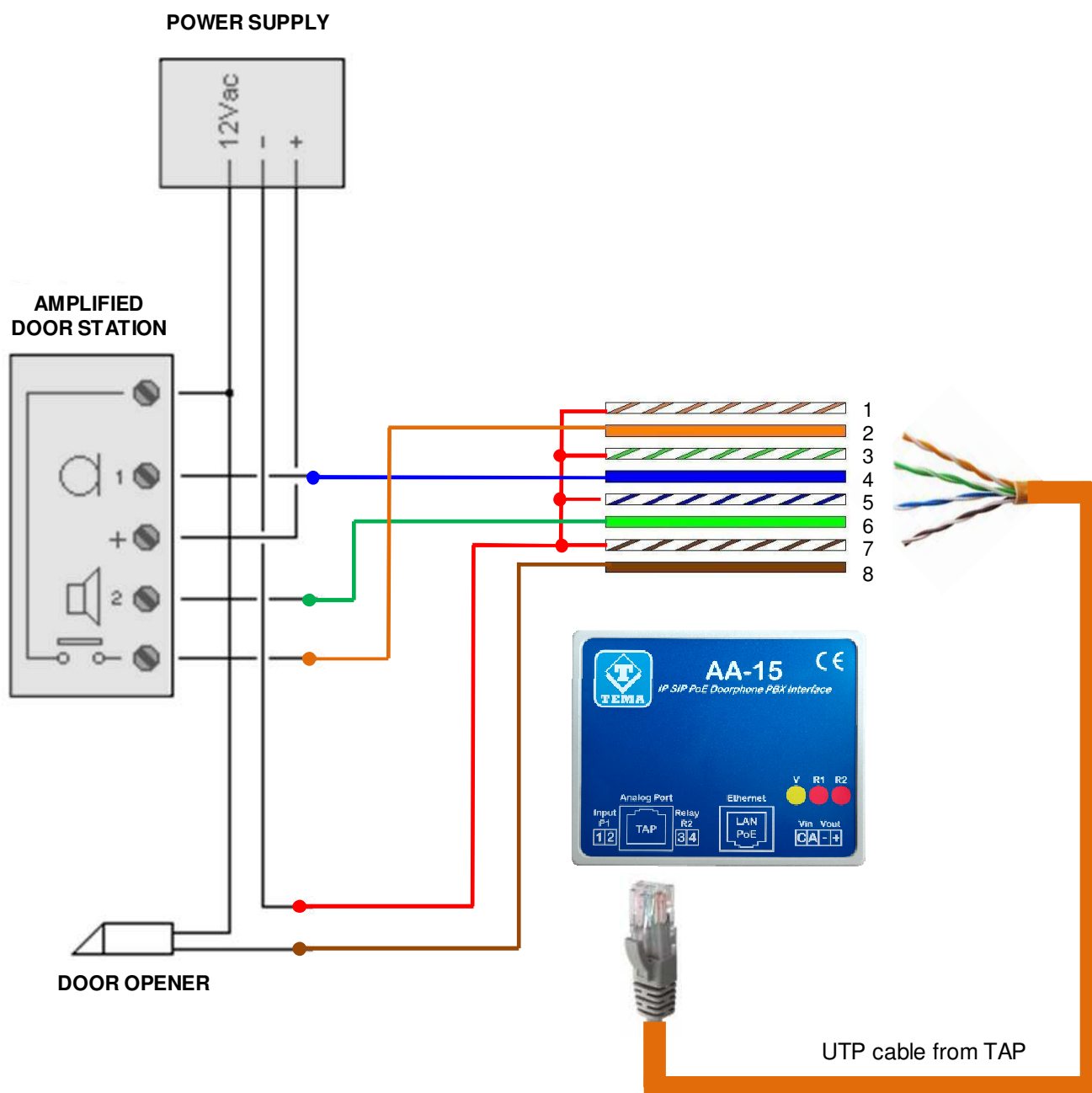
8.4. Connection with URMET mod. 1130 intercom



8.5. Connection with BPT intercom



8.6. Connection with TERRANEO intercom



8.7. Connection with FARFISA Intercom

Update in progress

8.8. Connection with COMELIT Intercom

Update in progress

8.9. FAQ Frequently Asked Questions

How is AA-15SIP powered?

AA-15SIP can be powered directly from the UTP LAN cable through a PoE switch, if not, it can be powered with a PoE injector or with an external power supply 230Vac/12Vdc, both in the Tema catalog.

Does AA-15SIP work with an Asterisk VoIP PBX?

Yes. AA-15SIP has also been tested with all the PBXs of the most prestigious brands such as: SIEMENS - AVAYA - ALCATEL – PANASONIC – SAMSUNG - NEC - 3CX - LG - WILDIX - AASTRA - ASCOM - NITSUKO - SELTA – PHILIPS - MITEL E SISTEMI BASATI SU ASTERISK, ... see list at page 5.

How it can be installed and put into service?

Few steps:

- 1) Connect AA-15SIP in a LAN socket to a PoE switch with standard cat. 5/6 cable, or to a generic switch by powering it with the external power supply.
- 2) Connect with a browser and assign an IP address and LAN credentials.
- 3) Register the number assigned to AA-15SIP in the IP-PBX switchboard in the space reserved for your SIP account or, in the absence of the IP-PBX, arrange for the doorphone call to be sent to a SIP phone in P2P mode (Peer-to-Peer).
- 4) Connect the UTP cable coming from the "TAP" socket of AA-15SIP to the points indicated in the connection diagrams provided for the various traditional intercoms on the market in a "parallel" way on the necessary signals.

When a visitor presses the button, AA-15SIP generates a SIP call to an internal number, communicating the visitor with the answering operator, who has the possibility of operating the relay for opening with a code from the telephone of the gate. In any case, the internal intercom station remains operational since AA-15SIP is transparent and works in parallel.

I can't get to the place where I have to install AA-15SIP with a UTP cable, can I use a Wi-Fi link?

Yes, AA-15SIP is a normal LAN network terminal, in this case you need a client access point with the RJ45 LAN output towards AA-15SIP and a power supply (for example the plug model T7012L or for DIN rail AA-39D1A) connected to a 230Vac mains socket in the immediate proximity.

Can I connect AA-15SIP on a 2-wire BUS intercom system?

NO, AA-15SIP currently only works on traditional 4-5 wire intercom systems. If this need exists, contact TEMA by supplying the brand and model of the intercom with 2-wire BUS operation to check whether any operating compatibility has been released on the date.

I have to install AA-15SIP on a 4-5 wire intercom system but can't find the diagram in the manual?

The compatibility of AA-15SIP with the various intercoms on the market is constantly evolving, contact TEMA providing the brand and model of the intercom to be adapted.

I installed AA-15SIP and everything works correctly, however there are some disturbances during communication

The quality of the audio received and transmitted by AA-15SIP is strictly dependent on the quality of the intercom system, the device does **NOT correct** any disturbances or low audio quality of the existing intercom system but transfers the audio exactly as it receives it. As they are old intercom systems, it is possible that the quality of speech is already compromised, **therefore communication must be considered acceptable**, even with some disturbance, provided that it is intelligible even at a minimum level of understanding. After all, long conversations are not normally made at the intercom unit.

NOTES