TEMA TELECOMUNICAZIONI

"IP COMMUNICATION AND SECURITY COMPANY"

WWW.TEMATLC.IT

AD630S / AD630SA

VoIP SIP 30W PoE Horn Speaker
IP66 grade of protection – 120dB – 2 Way

Audio Over IP Network Series

"SipComStage"

Products for VoIP SIP LAN networking Communications, Paging Amplifiers Systems and Audio Applications. Zone Announcements, Night Ringer, Multicast general call, Background music, Standard SIP call or Emergency call, SD memory with pre-recorded messages, Relays for LAN drives.

PRODUCT MANUAL

HW Version 1.0 – SW Version 1.1

CE

Revision	Date	Revision reason	Prepared	Checked/Approved
0	19/04/2022	First edition	MM, 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	**Horn VoIP SIP 30W PoE IP66**
Trade name	**TEMA TELECOMUNICAZIONI Srl**
Type or model	**AD630S, AD630SA**
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, 20 February 2022

TEMA TELECOMUNICAZIONI SRL
D. Pontillo

I. IMPORTANT INFORMATIONS REGARDING THE RECOVERY AND RECYCLING OF THIS ELECTRONIC DEVICE

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.

TEMA TELECOMUNICAZIONI is a certified company

UNI EN ISO 9001:2015

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.
10. This document is property of Tema Telecomunicazioni, It is forbidden any duplication and reproduction, even partial, as well as storage on any medium without written permission of Tema Telecomunicazioni.

This symbol in the descriptions indicates a general warning or a damage danger to equipment or people.

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

Useful references TEMA TELECOMUNICAZIONI

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Web: WWW.TEMATLC.IT WWW.TEMA-IPAUDIO.COM

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1. PRESENTATION

AD630S / AD630SA allow to realize Paging Amplification Audio Sound Systems in a standard LAN network completely integrating with IP-PBX SIP in order to be able to play announcements from any telephone in the internal network and any technologies: analog, SIP, Smartphones and Softphones.

Access can be authorized by Password with code to be typed before starting the announcement. **Two internal relays** can be programmed to be automatically activated when it is called or to be controlled from the phone via LAN that originated the call to the AD630S horn, eg. to report the event to any auxiliary devices connected downstream.

The system programming is done remotely via LAN thanks to the integrated Web server. AD630 incorporates an efficient 40W (2x20W) Class-D amplifier.

2nd SIP Account – Night Ringer. AD630S can be registered on the PBX with a second telephone number, different from the one used for ads and inserted in the night ring group. To the arrival of an incoming call, the system plays a sound (user selectable) to all the network speakers. The volume can be adjusted independently of the others managed audio channels.

Inputs from external contact with a variety of functions, for example: provide to make a SIP call to a preprogrammed number and inform the operator of the event with an appropriate voice message. The signaling can be repeated a number of programmable times, the operator can interrupt the sequence with a keyboard code. Other examples of use will follow in this manual.

Multicast Audio Streaming for music and announcements diffusion. AD630S handles up to 16 multicast channels in LAN with priority levels to allow the background music broadcasting on loudspeakers network. The generation of musical programs in streaming audio can be managed from a PC of the LAN/WAN network with special software or special TEMA Encoder.

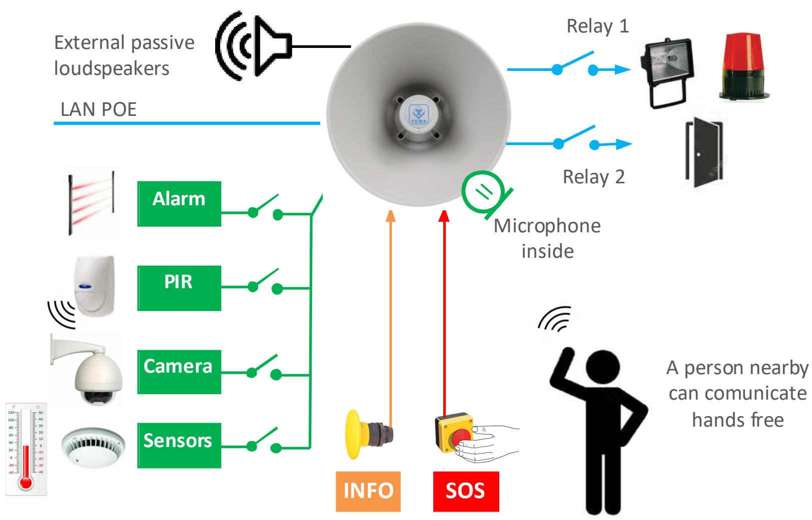
Service "SIP Security Info Call" - Emergency SIP Call. By connecting a button or an external contact, AD630S can call a phone number programmed to alert the event with a specific warning message. It is a security service for emergencies or simple request for help and information, and allows a one-way phone call.

Total remote management via LAN, integrated Web Server. Programming, configuration, loading and listening to audio files, firmware release update, audio volume adjustment, backup and restore the configuration, device reboot.

Possibility to connect one or more external passive loudspeakers on the **2nd amplifier** to expand audio coverage of the served area. High power output of 20W when powered with an external power supply, automatic lowering of the power with 10W limitation when powered only from LAN cable via PoE.

Input/Output

- 1 Multifunction Alarm input
- 2 Input push button for SIP calls or other 2 programmable inputs for Alarms
- 2 Remote controller Relays
- 1 Microphone inside
- 1 Power input 12-24Vdc
- 1 Power output for service 12-24Vdc
- 1 Output of the 2° inside power amplifier 20W to connect external passive loudspeakers



VoIP SIP Amplifiers-Speakers and accessories of the AD600 series

SipComStage

AD630S IP-SIP PoE 30W alluminium Horn Speaker with 40W internal amplifier (2x20W) IP65 protection. Same design **AD330/30T** passive horn 30W-8Ohm with transformer.

AD630SA IP-SIP PoE 30W ABS oval Horn Speaker with 40W internal amplifier (2x20W) IP65 protection. Same design **AD330A/30T** passive horn 30W-8Ohm with transformer.

AD635 IP-SIP 12W PoE Speaker, wall mounting. Same design **AD335/06TP** passive speaker 6W-8Ohm with transformer.

AD633 IP-SIP speaker 12W PoE ceiling mount version. Same design **AD333/12TP** passive 12W-8Ohm speaker with transformer.

AD634 IP-SIP 2x20W PoE IP65 Speaker in vandal-proof aluminum. Same design AD334/20TP passive 20W-8Ohm speaker with transformer.

AD638 IP-SIP 2x20W PoE IP65 speaker two-way low-medium and high. Same design **AD338** passive speaker 30W-8Ohm.

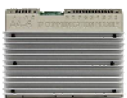
AD639 IP-SIP 2x20W PoE IP56 Speaker. Same design **AD339** passive speaker 15W-8Ohm.

RingOne

AD639SR IP SIP 30W Call Repeater, integrated Power Supply PoE 802.af, IP56 protection grade.

AD610 2X2W PoE IP-SIP Amplifier Module, External Amplifier Adapter with Sound Isolation Transformer. Directly pilot external power amplifiers or 8 Ohm external passive speakers.

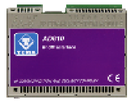
AD611 IP-SIP PoE 10W Amplifier Module (2X5W). Directly pilot 8 Ohm external passive speakers.



AD612 IP-SIP PoE 40W (2x20W) Amplifier Module when powered with an external power supply. Directly pilot 8 Ohm external passive speakers.

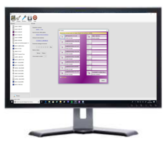
AD699/XPB Die cast Metal Box, IP65 protection grade, antivandal with red N.A button.

AD615/S Audio Encoder Module from analog to digital on LAN Network in Multicast/Broadcast Channels.



AD615/SIP Audio Encoder from analog to digital on LAN Network in Multicast / Broadcast Channels, **with SIP protocols** for calls routing to selectable Multicast channels.

AD696/BM External microphone base with "Talk" button and 7 selectable Chime for AD615/S RJ45 plug. Version **AD696/B** without chime.



ADAM
Audio Domain & Access Management
Management software and microphone console from 3 to 256 zones

AD696/AA External Microphone Base with "Talk" button specific for PC with ADAM software

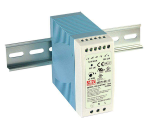
AD696/CT4 Mic Console 3-12 zones
AD696/CT8 Mic Console 7-28 zones
AD696/CT40 Espansion Module 40 zones

AA-39DL
Plug Power Supply 220Vac/12Vdc-1,0A.

AA-39D1A
DIN Power Supply 220Vac/15Vdc-1,0A.

AA-39P4
PoE injector 230Vac/48Vdc 0,5A.

AA-39E3
DIN Power Supply 220Vac/24Vdc-1,5A



AA-39E6V
DIN Power Supply 220Vac/24Vdc-2,5A.

AA-699/xxx Flash lights with Xeon lamp or high brightness LEDs. Ideal to be controlled by the relay of the IP-SIP speakers of the AD630 series or by the IP-SIP amplifier AD610, AD611, AD612 modules, with a call in progress add a light signal to the acoustic signal.

ANALOG COMPONENTS

AD301R 30W analog amplifier, 2 channels, Volume-High-Low +/-12dB controls

AD320/30 PA 100V audio line transformer 30W, constant voltage, high efficiency toroidal type with low magnetic flux leakage.

AD32 Isolation transformer, balanced, mixer, attenuator, audio signal filter, low impedance output. Ideal for decoupling devices of audio sources to other amplifiers to eliminate noises.

AA-36 Protected Power Relay Actuator 250Vac/16A/4KVA, command in AC/DC 12V/24V, 1 contact, Led, contacts protected by electric arcs. Ideal for driving loads up to 4.000W.

AA-39D2
DIN Power Supply 220Vac/15Vdc-2A-30W

AD330/xx Are available Speakers of different power, Horns and Loudspeakers for flush and ceiling mount, indoor and outdoor, IP54-65.



Unique features of the "SipComStage" AD600 Audio over IP line

- *SCALABILITY*
- *RELIABILITY*
- *FLEXIBILITY*
- *EASY TO USE*
- *EASY TO MANAGE*
- *REMOTE CONTROL*
- *HIGH SOUND QUALITY*
- *INTEGRATED POE POWER SUPPLY*
- *TERMINAL INTERCHANGEABILITY*
- *UNIQUE AND EXCLUSIVE PERFORMANCES*
- *EASY SOFTWARE UPGRADE*
- *CE CERTIFICATION*
- *LOW COSTS*
- *INTEGRABLE WITH ANY KIND OF IP-PBX AND SOFTCLIENT*
- *INTEGRATION WITH EVERY SAFETY SYSTEM*
- *LAUNCH OF ADS AND COMMANDS VIA GSM/UMTS*
- *2-WAY COMMUNICATIONS - INTEGRATED MICROPHONE*
- *PROTECTION FROM IP54 TO IP67*
- *VANDALPROOF VERSIONS FOR INDUSTRIAL ENVIRONMENTS*
- *CONTINUED UPDATES AND INNOVATIONS*
- *CURRENT ADVANCED TECHNOLOGY*

Compatibility with the most common brand of IP-PBX:
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 – NETHESIS - 3CX - KALLIOPE – BRIA – ASTERISK BASED SYSTEMS AND SOFTPHONE APP – DOMOTIC SYSTEMS LOXONE

***MADE IN ITALY**: all the products are designed and manufactured in Italy by TEMA which guarantees their support and technical assistance over time with a 24 months warranty.*

2. FEATURES

- **AD630S** IP PoE Horn with aluminum alloy cone, **AD630SA** IP PoE ABS Horn, IP65 grade of protection
- Dimensions 245 x 245 x 300 mm (AD630S), 215 x 283 x 290 mm (AD630SA)
- Montaggio a parete, soffitto, in interno/esterno, uffici, magazzini, capannoni, ecc..
- Wall mounting, ceiling mounting, indoor/outdoor sheds, offices, warehouses, etc..
- PoE (Power over Ethernet) 802.af connected via UTP cable or external power supply 12-24Vdc, even simultaneously with both power supplies
- Audio output power **40W (2x20W)** with 24V external power supply, 10W (2x5W) with PoE power
- **2° SIP account** adjustable on the IP-PBX for Night Ringer mode with the possibility of routing the audio content of the ringtone file to a programmable multicast channel.
- **16 RX Multicast Channels** for Music/Ads/Messages diffusion with 7 priority levels
- **9 Multicast Channels in trasmission/Routing**
- **2 remote controlled relays** via LAN
- **1 multifunction input** from external dry contact
- **2 buttons inputs** for direct call to SIP numbers differentiated by communication/alert
- **1-2 pre-recorded announcement**, associated with the available input, or up to **9 announcements** by making a call and sending the respective code via telephone
- Internal memory for customizable audio files (max 60s each)
- Possibility of diffusing pre-recorded announcements at pre-established times (up to 5 for each day of the week), from table or up to 32 per day with annual calendar programming
- Possibility of diffusing a pre-recorded daily announcement, at a set time or manually controlled, with programmable interval and repetitions
- **Notification service** with dedicated messages following external event
- “SIP Security Info Call” warning/emergency telephone call, sending of pre-recorded messages on loudspeakers and on the called
- "Push to Talk" function to control the direction of communication from an external button
- Independent adjustment of audio volumes: communications, multicast, night ringtone, alert tones
- VoIP connection with standard SIP Proxy Server protocol
- P2P (Peer to Peer) for operation also without IP-PBX
- Free APP developed for smartphones/tablets with iOS and Android operating systems
- AA-Videoconsole software for Windows XP/7/10 PC for management of TEMA IP terminals included
- ADAM Software (Audio Domain & Access Management) for Windows 7/10 PC for total management of the system
- Programming via dedicated Web Interface with password protection
- Possibility of software / firmware update via LAN
- Possibility of receiving and sending MULTICAST RTP audio streaming, for paging audio announcements, up to 7 priority levels (in reception) and with different volumes for each channel
- Routing of the audio received in a SIP call to a programmable multicast destination
- Stream of the alarm message to a programmable multicast destination
- Audio streaming with high quality codecs
- Autotest automatic Audio for functionality check, alert via email / SIP call in case of anomaly

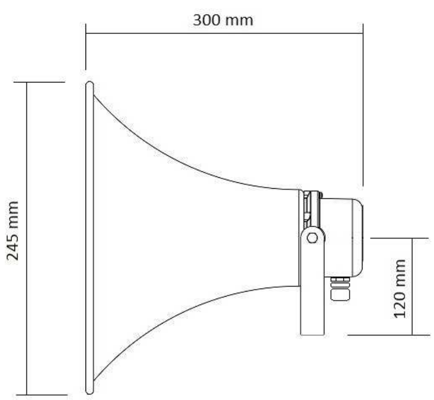
3. PACKING LIST

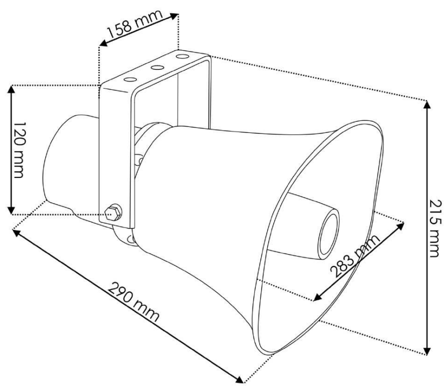
The systems are supplied with the parts included in the following list:

- An IP horn speaker device cod. AD630S / AD630SA
- A 3mt LAN cable connected internally to the horn for pre-installation test functions
- A printed system documentation, installation, programming, user manual
- A CD-ROM with complete documentation

4. TECHNICAL SPECIFICATIONS

LAN	TCP/IP Network 10/100BaseTx
Protocols	SIP 2.0, RFC 3261
RTP Multicast audio Streaming	G711µ, G711a, G722, L16 from 16 to 32 kHz
Connection	SIP Server (IP-PBX) or P2P (Peer To Peer)
Power Supply	PoE, Injector PoE and/or external Ac/Dc power supply
PoE	802.3af class 0 12,95W
External Power Supply (Opt.)	230Vac / 12-15Vdc -1A, 24Vdc - 2A
Technology	MIPS 560MHz Microprocessor
Memory	128MB Ram, Flash 32MB
Programming	practical Web interface and password
APP for Smartphones	iOS, Android
Messagges/Sounds	prerecorded in the internal SD memory card
Audio File Format	Windows .wav – 8K – 16 bit
Duration	60 sec
Bandwidth	300Hz – 7KHz
Power	Class-D 2x5W (PoE) amplifier, 2x20W (Ext. Power Supply)
Sound pressure	120dB A (SPL) max power
Dispersion angle	100° @ 2KHz
Audio communication	Unidirectional / Bidirectional / Push to Talk
Internal Microphone	Omni-directional electret 30Hz-18KHz
Additional speaker output	1 output for external speaker, passive, impedance 8 Ohm
Automatic check mode	Through self audio feedback
Inputs from buttons or external contacts	3 for buttons or warning/alarm
Integrated relays	2
Relay contacts capacity	Max 30Vdc – 1,5A
Signalings	Active call Led, call tone, Ding-dong ads
Installation	Ceiling mount, wall mount
Housing Material	Aluminium and ABS
Grade of protection	IP66, weather resistant
Storage temperature	From -20° to +65°C
Operating temperature	From -20° to +60°C
Relative humidity	Up to 100%
Dimensions	245 x 245 x 300mm (AD630S) / 215x283x290 (AD630SA)
Weight	2,1 Kg (AD630S) / 2,2 Kg (AD630SA)
Warranty	2 years, possibility of extension 3 to 5 years (Option)
Compatibility	CE, ROHS





AD630S
Aluminium round, grey

AD630SA
ABS oval, white

5. OPERATION

In stand-by and correctly configured, the system waits for incoming calls or active Multicast channel.

The device allows also the “streaming” Multicast sound diffusion. In this case the RTP audio diffused in LAN will be played amplified by the AD630S horn. Differing from the announce mode on direct call, with multicast mode, more devices can diffuse the same message at the same time, useful for general pre-recorded ads or for music sound diffusion.

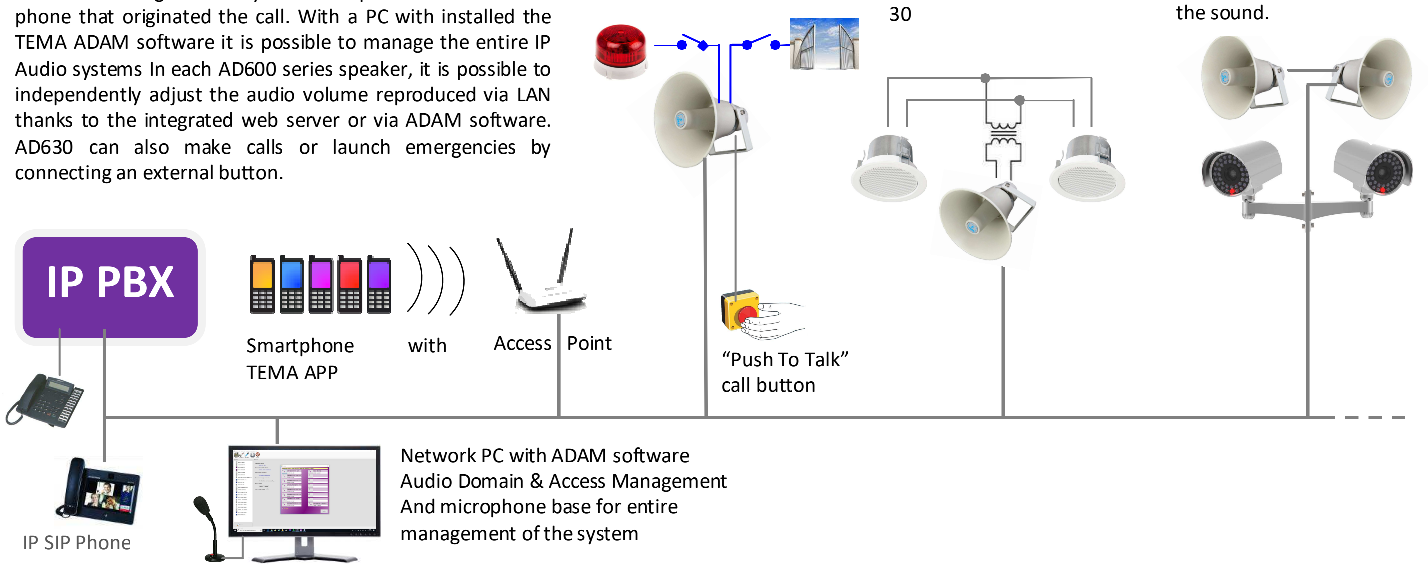
The operator who want to make an announcement on the device will make a call, from any extension of the PBX, to the number to which the device is registered or to its IP address in the case of Peer-to-Peer connection.

It is always monitored the change status of an alarm input to which it is possible to associate a number to call and a message to be played to the called party answer. It is possible to make common adjustment of the level output of the speakers to better suit the acoustic characteristics of the served area.

5.1. Device connection diagrams

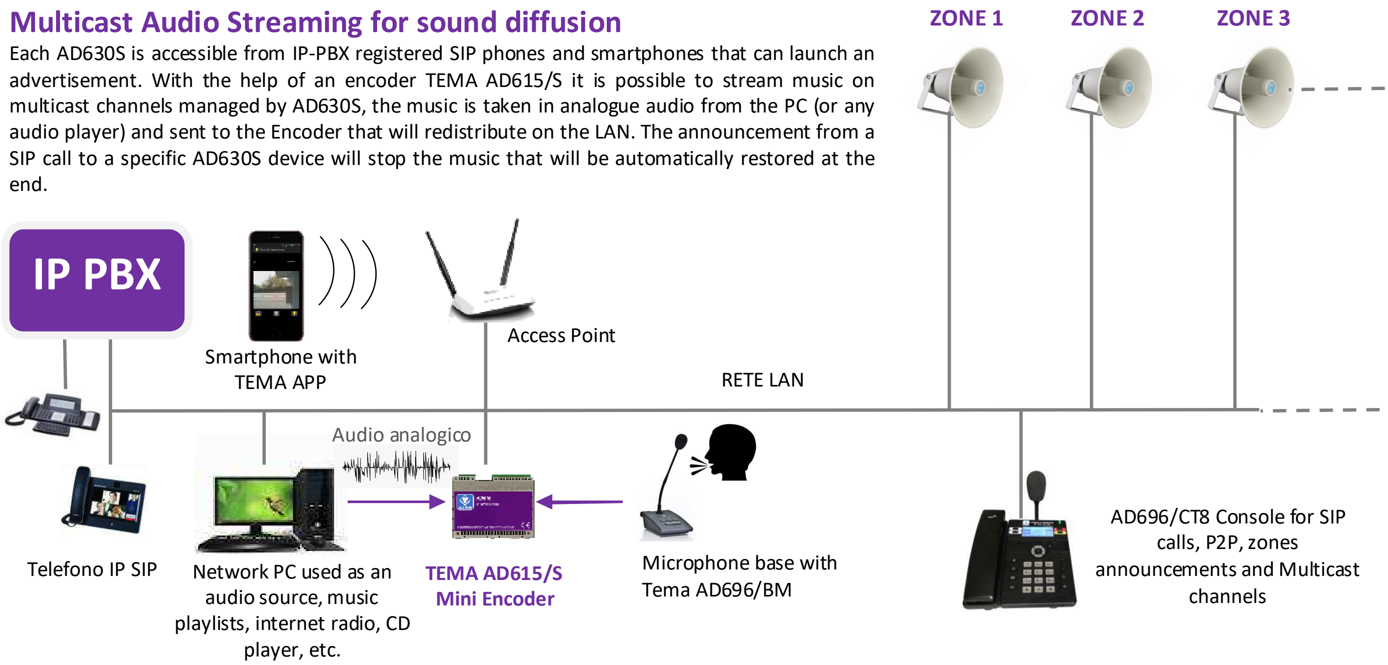
Application with IP-PBX or Peer to Peer

Each IP Horn can be reached from SIP phones, Smart-phone and other registered terminals in the IP-PBX that can launch ads. The 2 integrated relays can be operated from the SIP phone that originated the call. With a PC with installed the TEMA ADAM software it is possible to manage the entire IP Audio systems In each AD600 series speaker, it is possible to independently adjust the audio volume reproduced via LAN thanks to the integrated web server or via ADAM software. AD630 can also make calls or launch emergencies by connecting an external button.



Multicast Audio Streaming for sound diffusion

Each AD630S is accessible from IP-PBX registered SIP phones and smartphones that can launch an advertisement. With the help of an encoder TEMA AD615/S it is possible to stream music on multicast channels managed by AD630S, the music is taken in analogue audio from the PC (or any audio player) and sent to the Encoder that will redistribute on the LAN. The announcement from a SIP call to a specific AD630S device will stop the music that will be automatically restored at the end.



6. OPERATING MODES

6.1. Calls to devices

Call from the PBX phones to the AD630S system allows to make announcements to the network speaker. However, it is a normal telephone call, typically unidirectional although AD630S makes possible to hear the sound present in its vicinity, being equipped with an internal microphone. Calling the extension number connected to the systems, they will answer after a set time. It is also possible to have an appropriate code to protect the function from unauthorized access. Hang up to end the announcement. Any background music will be automatically restored.

6.2. RELAY function

The relays are typically used to signal to any auxiliary external devices that the announce is played, automatic activation call per call. If not used for the function described above it is available and can be controlled with an appropriate code from the caller's telephone.

6.3. Calls generated for acquisition of external contacts for special signalings

Calls that the system generates on his line to alert after triggering the alarm by the equipment input contact connected (aux devices or a button available to users in the proximity of the device). At the answer it is played to the called a pre-recorded message. It is possible to assign a phone number or an IP address that will be called when the input is activated. It is possible to determine the activation status of the same. It is the closure of the external contact that triggers the sequence of signaling but it is also possible to reverse this logic, in this way the signaling can take place after the opening of the contact (normally closed) connected to the input.

AD630S continuously monitors the status of the contact in the event of the condition of the activation stored on non-volatile memory card. As soon as possible it will begin to call the person who will manage the situation found at the telephone number programmed reproducing the message associated with the event. It is also possible to choose to play on a local speaker and on any connected audio output a pre-recorded message, which will be played prior to the call to alert the present persons in the area served by the system detected event. It is possible to define an acquisition/silencing code of alarm signaling that the called must type to inform the device of the alarm acknowledgment. If the called number is busy or does not reply, or in any other case where AD630 does not receive the capture/acknowledgment code, at the end of each call attempt return in standby and prepares itself for a new alarm notification.

When the systems receive the correct acquisition/acknowledgment code related to the alarm in progress, the alter signaling will stop and will not be carried out further alarm notification calls. In order to be ready again, it is necessary that the condition which had triggered the previous notification returns in standby. Only at the occurrence of a new activation of the contact condition the calls cycle with the alert notification will start again. In practice: if a connected contact is closed and it is detected its activation, AD630S starts to make warning calls.

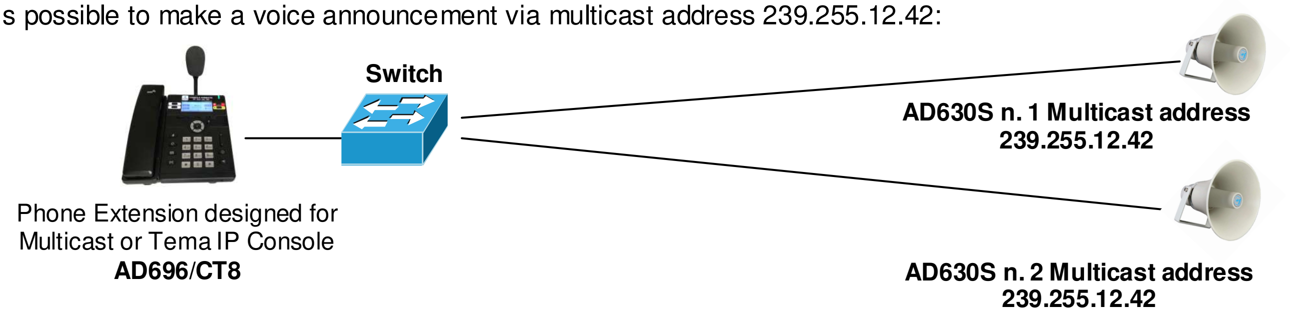
In the event that the person called by the system answers and successfully enter the acknowledge code, the alert calls will be terminated. If the contact that had triggered the alert, however, remained closed (this obviously depends by the external device that controls the contact) will NOT be triggered another warning cycle! To have new alerts for this contact, it is necessary that come back in standby before (reopening) and then to its new next possible closure will be detected from AD630S system with new telephone alerts.

Finally, it is possible to send the alarm message in stream to a multicast address, so that it can also be reproduced on remote Tema IP speakers.

6.4. Multicast Audio Streaming

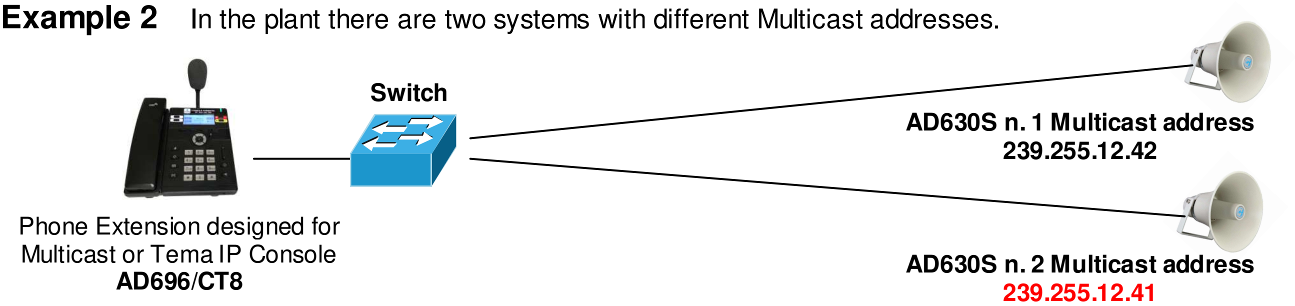
In a LAN network, the Multicast term indicates the possibility to distribute an information like data, audio or video, to a group of IP terminals of the Ethernet network. For Multicast are used class D IP addresses and they range from 224.x.x.x to 239.x.x.x. In our case the AD630S system is able to receive audio in Multicast into the specific IP channel and play it amplified. The audio can be generated from a PC software application (for example VLC) or by a IP SIP phone. The received audio stream is played without the intervention of any operator, this function is useful for playing back voice messages and information or emergency announcements (also called "Paging" mode). Obviously the Multicast audio streaming is unidirectional, the audio stream is sent from source to destination, but not vice-versa. The audio stream can be sent simultaneously to multiple terminals (same IP Multicast), or only to distinct terminals (each provided with its own IP address). It is possible to program up to 16 audio receive addresses, each with its own priority and its playback volume, so that the same terminal, if it is already playing audio (eg music) can be interrupted by a higher priority stream (for example an announcement) and with a different volume (typically an announcement must be played at a higher volume than background music). The audio supported by this mode is in the G.711 format (aLaw or µLaw), G722 or linear high quality 16 bit (proprietary format, diffused through the Tema AD615/S Encoder or an Intercom AA-500/AA-600 Series).

Example 1 In a plant, there are two systems, connected to the same network switch. From a suitable phone, it is possible to make a voice announcement via multicast address 239.255.12.42:



Both systems will play the announcement audio message. Many more devices than those given in the example can be connected, since there are no limits.

Example 2 In the plant there are two systems with different Multicast addresses.



The phone, selecting to send the multicast voice to one or other address of the terminal, will decide which device must play the streaming message.

7. COMMANDS AND CODES

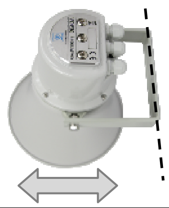
During the announcement will be possible to send the commands described here (which will be these by default):

DTMF Command	Function	Description
#1 - #2	**Activate the relay contact 1 or 2**	Used to manually activate the relays (if not programmed to be automatically activated per announcement).
*9x	**Announcement**	Follow the code *9 with the number of the announcement you want to play (1 to 9). Example: make a call, wait for the horn to answer, dial *91, the horn hangs up the call and emits the pre-recorded announcement file no. 1

8. INSTALLATION OF THE DEVICE

8.1. Mounting and positioning of the AD630S / 630SA horn

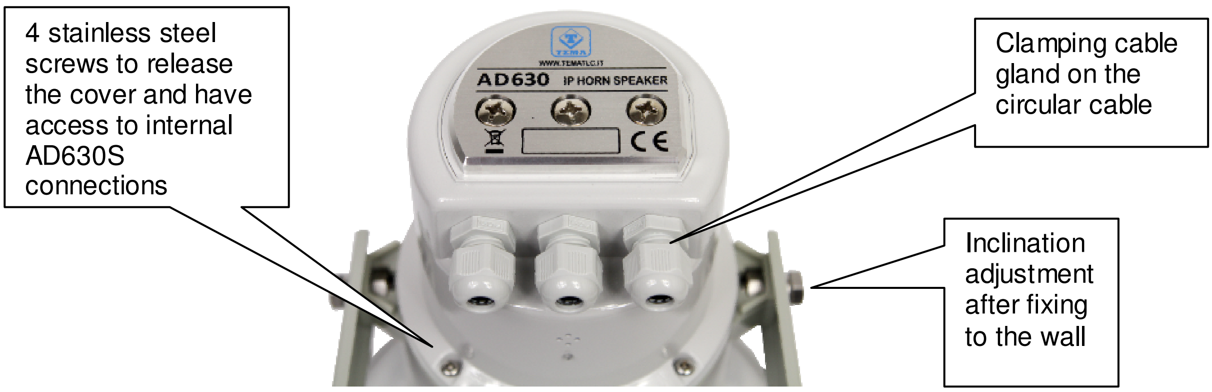
The outdoor horn must be fixed to a stable and suitable surface to support its weight also considering the fact that by hanging at the side or below the mounting surface, the force that the support material must oppose is greater than the nominal weight of the single unit to sustain.

Ceiling Mount	Right side wall mount	Left side wall mount	Roof mount
			
The pictures suggest the possibility of installing of the fixing bar and the subsequent possible orientation of the horn cone. We remind that the angle of sound diffusion of AD630 is a cone of about 100°.			

The wall mounting bar of AD630S has three holes. A hole is located in the center and two others are adjacent to 62mm distance from the central one. Each of the three holes has a diameter of 10mm, definitely suitable for use of generous screws/dowels. Given the variety of possible fixing surfaces of AD630S, it is not supplied any accessory for fixing. According to the cases and the surface material, use the most suitable material.

8.2. Opening of the cover and access to internal parts - passage of cables

Once the AD630S horn is fixed, proceed to its connection. Before it can be performed you must have access to the inside of the device by removing the four fixing screws of the cover and gently lifting it (in fact internally is connected by wires to the horn previously fixed to the wall). If necessary, to make wiring, detach the fastons that connect the audio driver horn (see the magnet) to the amplifier module that physically remains fixed to the cover.



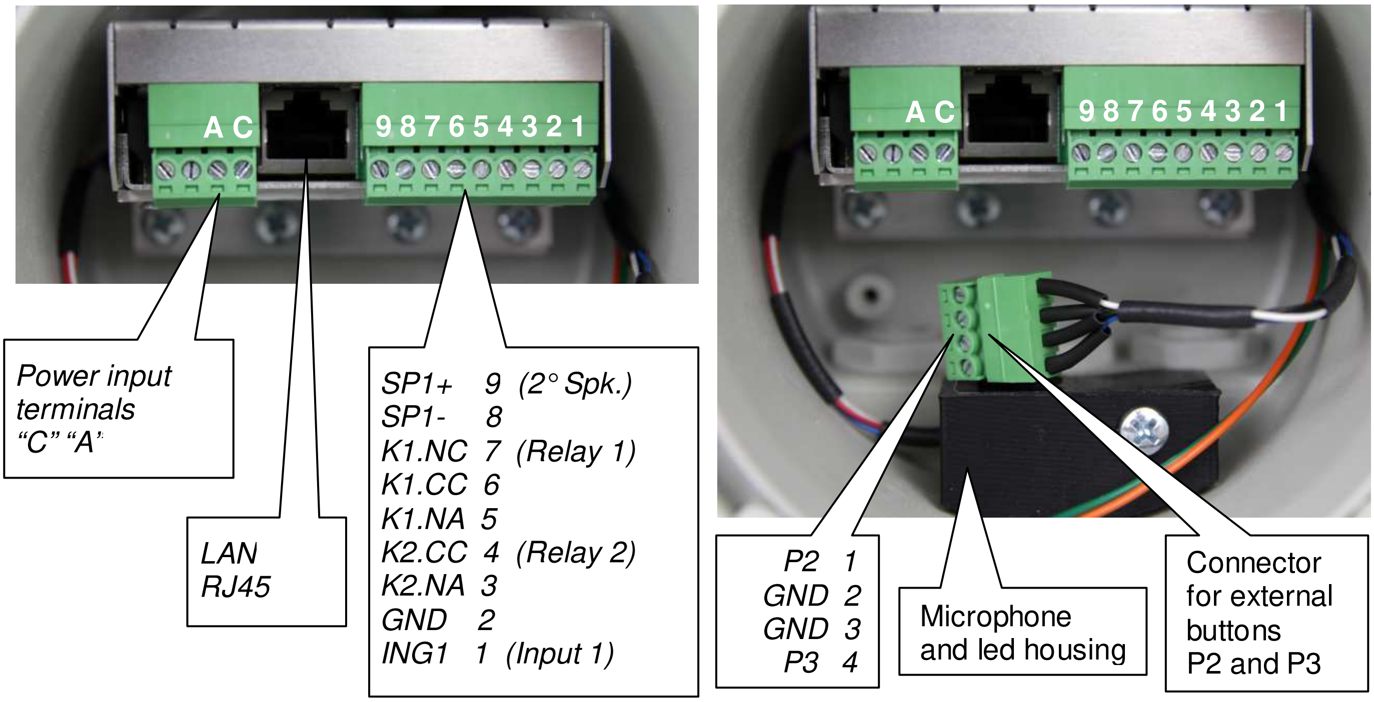
There is a polarity to be respected by reconnecting the two wires with the faston to the magnet winding, the orange wire is to be connected to the positive, marked with a red band on the magnet itself. The green wire is to be connected to the negative.

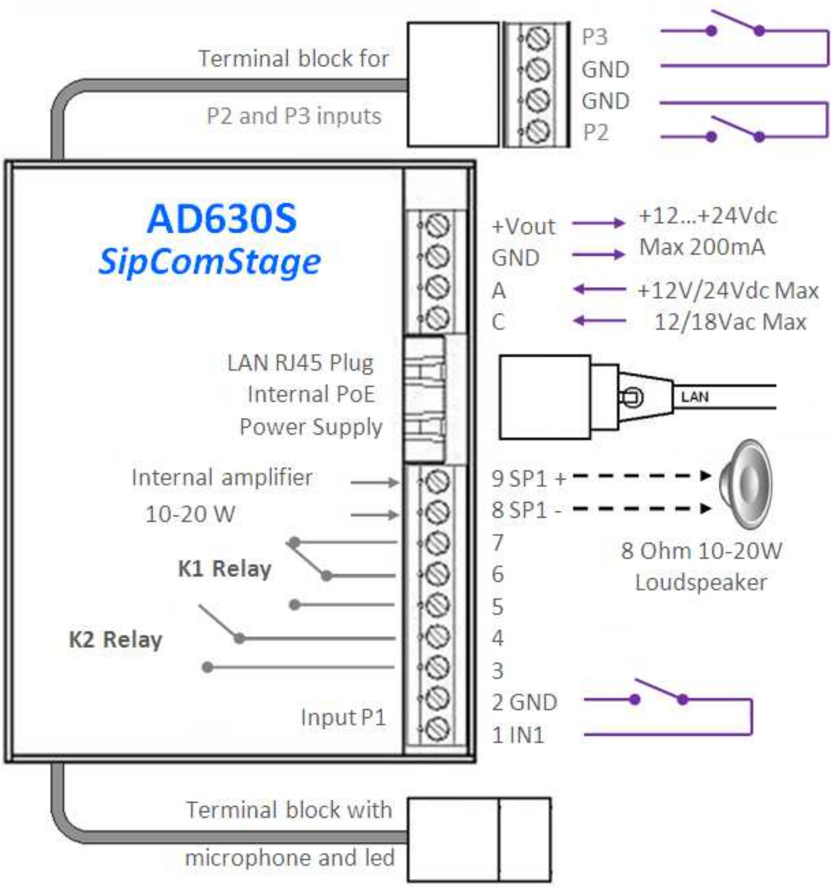
When the cover will be reassembled at the end of the connecting operations described later, remember to orient it so that it has always three cable glands facing downwards to avoid that any drips collected by the cables can reach the cable glands and infiltrate the cover or the horn.

The AD630S cable glands provide the best seal against external agents with the use of a round sheath cables of the sections from 4 to 8 mm maximum. Then use, according to the devices to be connected, up to three cables with suitable conductors and characteristics, a non-exhaustive example is: 1 LAN UTP-Cat5 cable, 1 power cable with two poles, a two-pole cable for an external additional speaker. Close each cable glands possibly not used for connections with a section of cable or other suitable expedient.

8.3. Connecting the internal module of AD630S

Inside of AD630S there is an electronic module for the management of communication functions on the VoIP network, and for amplification of the audio signal. The connection of the signal cables from the field must be carried out by qualified technical personnel, operate verifying that no cable is under voltage, also strictly comply with the instructions of the voltage and current values for the power supplies and other parts of the system. Follow the instructions below in the drawing block for the correct electrical connection of the system. On the following page there is the detail of each terminal shown.





4-pin removable terminal block (screw connector inserted to the electronic module)

+Vout	Terminal block from which it is possible to withdraw unregulated power supply, POSITIVE
GND	Terminal block from which it is possible to withdraw unregulated power supply, NEGATIVE
A	Terminal block to power the system, irrelevant polarity
C	Terminal block to power the system, irrelevant polarity

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..

RJ45 LAN port (present on the electronic module)

AD630S requires a LAN cable to connect to the network. If the port cable has also PoE, it will not be necessary to supply the module with other power sources (up to max 10W powers). For higher powers will be necessary to supply the module with other power source of 24 Vdc 2A or max 18Vac 2.5A.

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.

9-pin removable terminal block (screw connector inserted to the electronic module)

9 SP1+	Class-D amplifier 1st output terminal block for connection of a speaker, positive pole	
8 SP1-	Class-D amplifier 1st output terminal block for connection of a speaker, negative pole	
7 K1.NC	Terminal block of the K1 relay - NC contact, normally closed	
6 K1.CC	Terminal block of the K1 relay - CC contact, central contact	(contact load max 30V 1.5A)
5 K1.NA	Terminal block of the K1 relay - NO contact, normally open	
4 K2.CC	Terminal block of the K2 relay - CC contact, central contact	(contact load max 30V 1.5A)
3 K2.NA	Terminal block of the K2 relay - NO contact, normally open	
2 GND	Terminal block referred to the negative of the system power supply, for the input contact 1	
1 ING1	Terminal block for the detection of the input contact 1	

To terminals 1 (INP1) and 2 (GND) must only be connected a relay contact or a button free from any voltage to prevent permanent damage to the device.

4-pin removable terminal block (connector inserted to the electronic module)

P2	Connect here the dry contact of the button named P2
GND	Reference terminal of the P2 contact
GND	Reference terminal of the P3 contact
P3	Connect here the dry contact of the button named P3

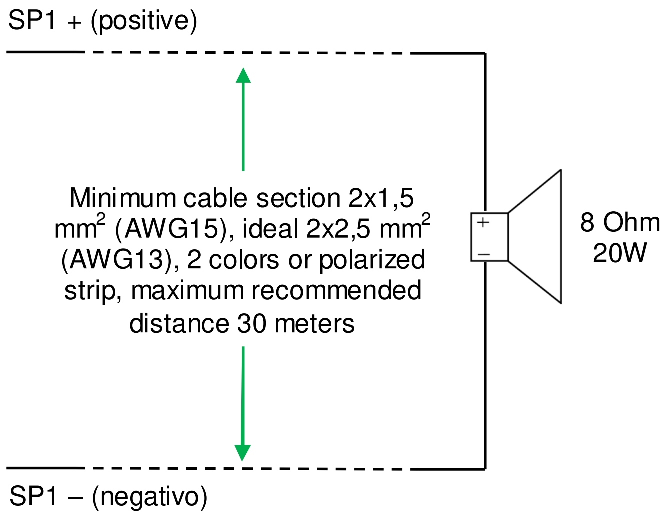
To terminals 1 (P2) and 2 (GND) / 4 (P3) and 3 (GND) must only be connected a relay contact or a button free from any voltage to prevent permanent damage to the device.

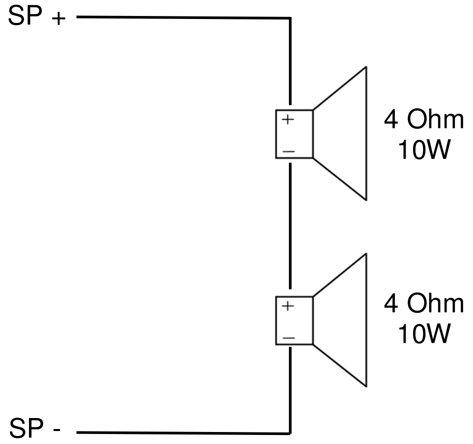
8.4. Correct connection of an additional passive speaker

The power amplified SP1 is an output where it is possible to connect an external additional speaker passive paying attention to its polarity but especially to the load impedance which cannot be lower than 8 Ohm.

Furthermore, it must be respected the maximum output power from the amplifiers using adequate power speakers. In the simplest case just connect an 8 ohm speaker of at least 20W power or higher. Use cables with different colors, section of at least 1.5 mm2 and limit the distance from the amplifier to no more than 30m.

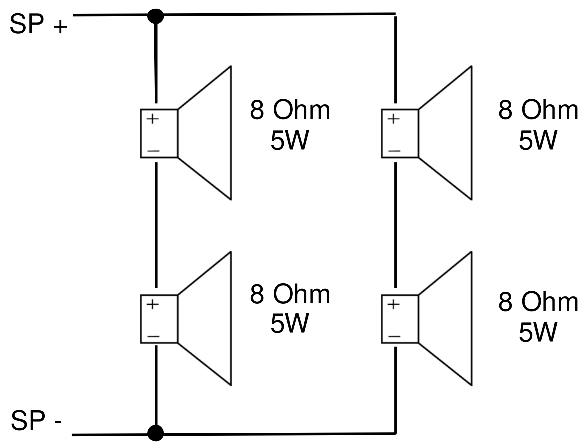
Below are some examples of mixed combinations, to serve a wider area but with distributed power
.





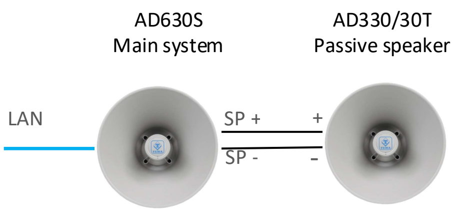
Corrispondenza cavi AWG / mm^2

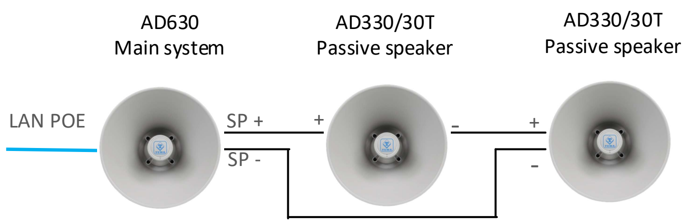
AWG20 >	0,518 mm^2	resistance 33,31 Ohm/Km
AWG19 >	0,653 mm^2	resistance 26,42 Ohm/Km
AWG18 >	0,823 mm^2	resistance 20,95 Ohm/Km
AWG17 >	1,04 mm^2	resistance 16,61 Ohm/Km
AWG16 >	1,31 mm^2	resistance 13,17 Ohm/Km
AWG15 >	1,65 mm^2	resistance 10,45 Ohm/Km
AWG14 >	2,08 mm^2	resistance 8,286 Ohm/Km
AWG13 >	2,62 mm^2	resistance 6,571 Ohm/Km
AWG12 >	3,31 mm^2	resistance 5,211 Ohm/Km
AWG11 >	4,17 mm^2	resistance 4,132 Ohm/Km
AWG10 >	5,26 mm^2	resistance 3,277 Ohm/Km



AD630S with additional speakers of the same design

Thanks to the 2nd internal audio amplifier it is possible to connect to the active IP unit another 1 or 2 low-cost passive speakers of the same design to increase power, directionality, extend the area served, even in different rooms separated by walls, in order to reduce the cost of the plant.

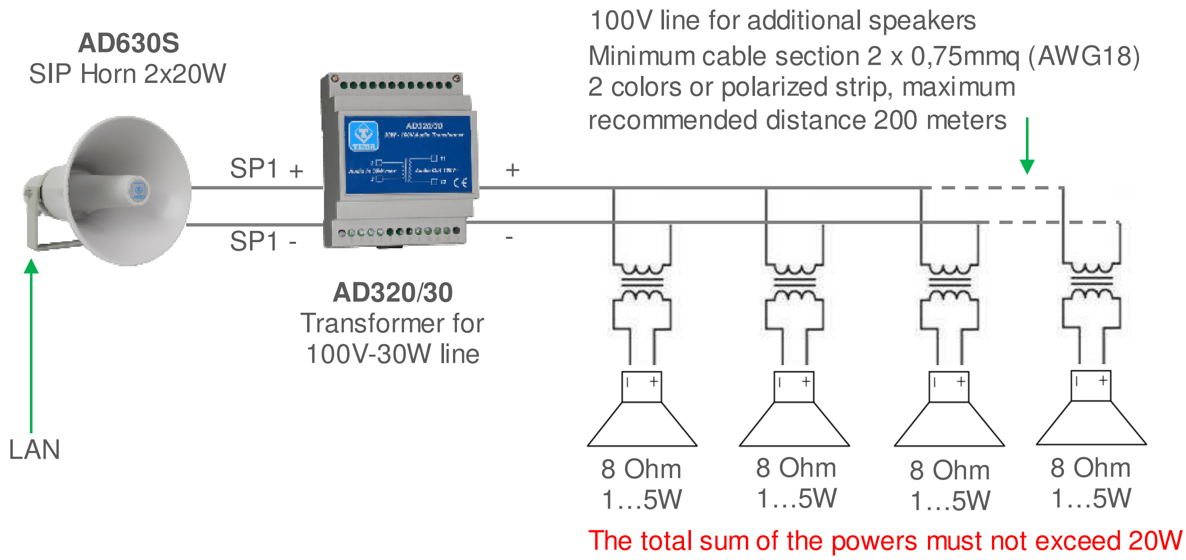




ATTENTION: when connecting an additional passive speaker it is necessary to power the active horn with the appropriate external power supply at 220Vac / 24Vdc-1.5A Tema AA-39E3 code or the Tema AA-39P4 injector code.

8.5. 100V audio line for additional speakers away from AD630S IP Horn

For special applications it is possible to generate from AD630S an audio line at a constant voltage of 100V of max 20W power using a Tema AD320/30 transformer to connect on the SP1 power output. In this way it is possible to install several other additional speakers (with internal transformer suitable for 100V line and selectable power) even at distances up to 200 meters from AD630S. The total sum of the power the installed speakers must not exceed the maximum power of 20W.



Some examples of speakers available on Tema catalog (not amplified, passive models)

	AD330/15T	Horn Speaker 15W / 8Ohm with transf. 100V, sockets 15/7.5/3.7/1.9 W response 300Hz - 7KHz, Dim. Diam. 210 x 240 mm
	AD330/30T	Horn Speaker 25W / 8Ohm with transf. 100V, sockets 25/12.5/6.25/3.12 W response 300Hz - 7KHz, Dim. Diam. 240 x 290 mm
	AD335/06TP	Wall Speaker 6W / 8Ohm Transf. 100V sockets 6/3/1.5 W response 180Hz - 16KHz, Dim. 218 x 216 x 120 mm
	AD334/20TP	Vandal proof Projector 20W with transf. 100V, outdoor, sockets 20/10/5 W, IP65 response 150Hz - 15KHz, Dim. Diam. 180 x 145 mm, weight 2.4kg, grey color
	AD337/06TP	Oval wall speaker 6W / 8Ohm with transf. 100V sockets 6/3/1.5W response 180Hz - 10KHz, Dim. 258 x 169 x 72 mm, 0,8Kg
	AD333/12TP	Round speaker wall/ceiling flush mounting, 12W with Transformer 12/6/3 W response 80Hz - 15KHz, Dim. Diam. 200 x 62 mm (Hole 160-165mm)

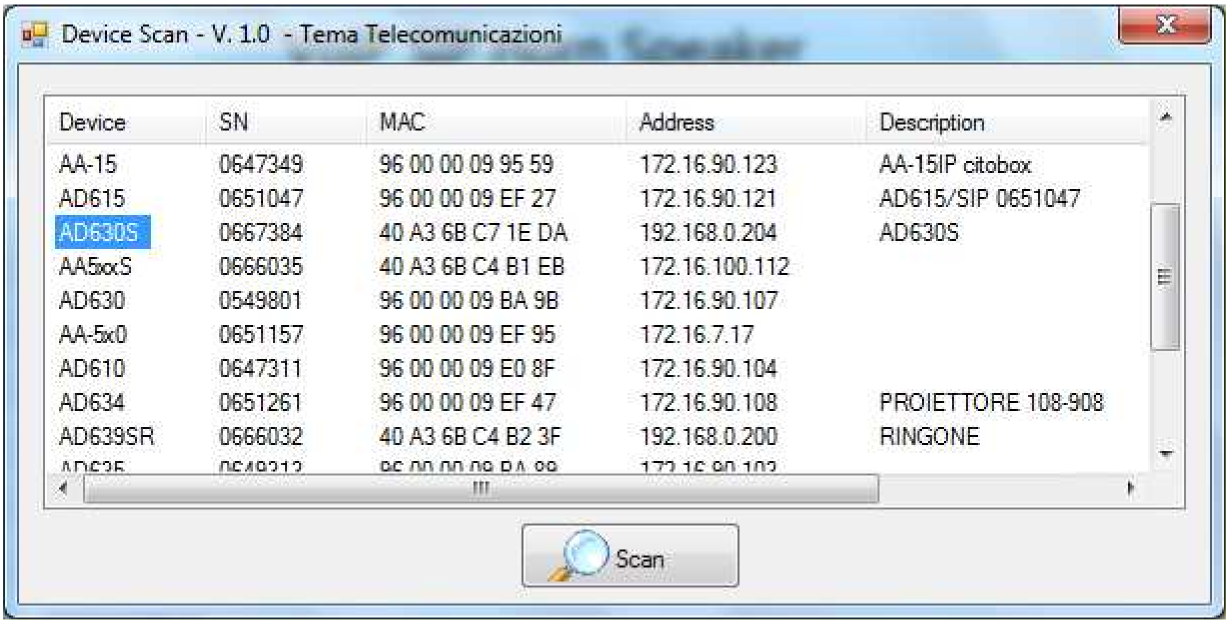
The speakers listed above are not produced by TEMA, but a selection of the best third-parties quality producers that are best suited to our production amplifiers and PA analog speaker systems and IP VoIP SIP for audio LAN. The components offered however fall into the corporate policies in terms of warranty and technical support.

9. PROGRAMMING

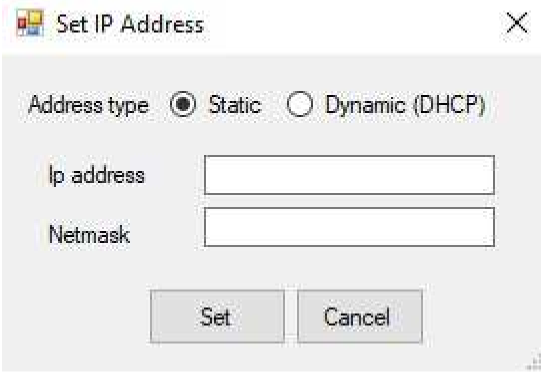
9.1. Preparation for programming the system parameters

Programming takes place through the WEB interface. To access, simply connect an Ethernet cable from a PC or switch to the LAN port of the AD630S.

The system is supplied with active 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 scan program for TEMA devices on the network 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 the 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.

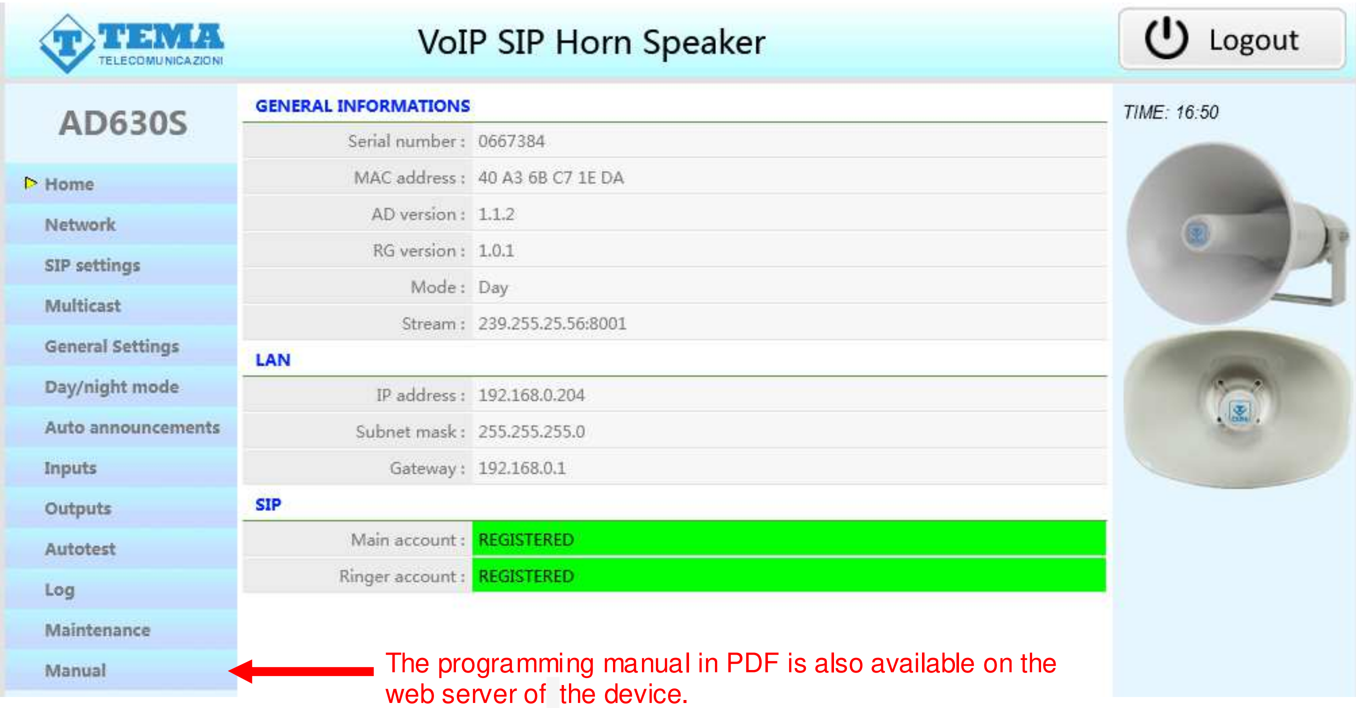
Programming takes place through the WEB interface. To access, simply connect an Ethernet cable from a PC or a switch to the LAN port of the AD630S.

9.2. Access to programming

To program the device, it is sufficient to use a normal browser, such as Explorer, Firefox, Chrome or other. The user / password to connect with are **master/master**. Once logged in, it will be possible to change the administration password for maximum security of your device, see par. 1.9.

Use is very simple and intuitive, the menu for choosing the functions to be programmed is always visible on the left, while the configuration mask active at that time is shown on the right. Any changes will be confirmed with the **"Save"** button. Closing the browser or changing the page without selecting this button **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:

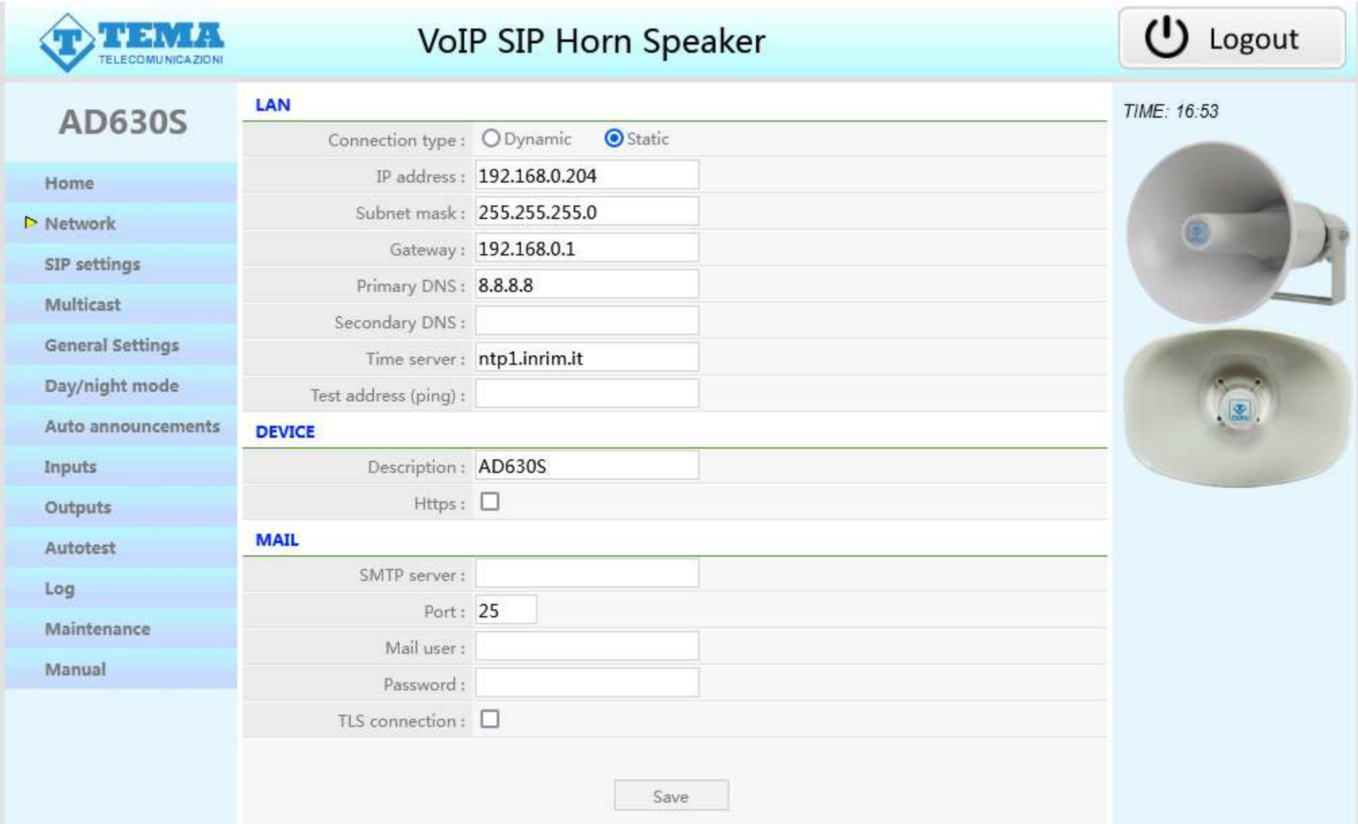


Any changes do not require the device to be restarted (except for changing the IP address and updating the software).

The address that appears next to the "Stream" field and the one programmed as # 11 on the Multicast web page (see below). It serves only as a reminder and facilitates the identification of the audio address assigned to each horn.

9.3. Network parameters

On this page you can set the network parameters, such as the IP address, the netmask, etc.:



If you choose the static network configuration, gateway and DNS are only necessary if you want the device to be able to access the Internet (for example to obtain the current date / time, in the example from the site ntp1.inrim.it).

Changes to the network settings are taken over by the device only at the next reboot. Once the configuration operations have been completed, therefore, remember to restart the system and possibly 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.
- *IP Address*: indicate the IP address you want to assign to AD630S.
- *Subnet mask*: indicate the network mask you want to assign to the AD630S.
- *Default gateway*: indicate the gateway that can allow AD630S to access the internet.
- *Primary and Secondary DNS Server:* Specify the addresses of the two DNS servers that you want to use to resolve the IP addresses.
- *Time server*: indicate the address of the server you want to use to have the time synchronization of the device.
- *Test address (ping):* if entered, this address is used by the system to verify the correct functioning of the network connection. It is advisable to enter the address of your pbx, first checking that it responds to ping requests.
- *Device description:* text that will appear on the search mask of the Adam software

It is possible to send an email when one of the inputs of the device changes state. Therefore, you need to set the characteristics of your account for sending below:

- *SMTP Server*: name or IP address of the e-mail server.
- *Port*: communication port of the mail server (usually 25)
- *Mail user*: user authenticated on the SMTP server
- *Password*: SMTP authentication password
- *TLS connection:* allows you to activate the secure connection for sending the email.

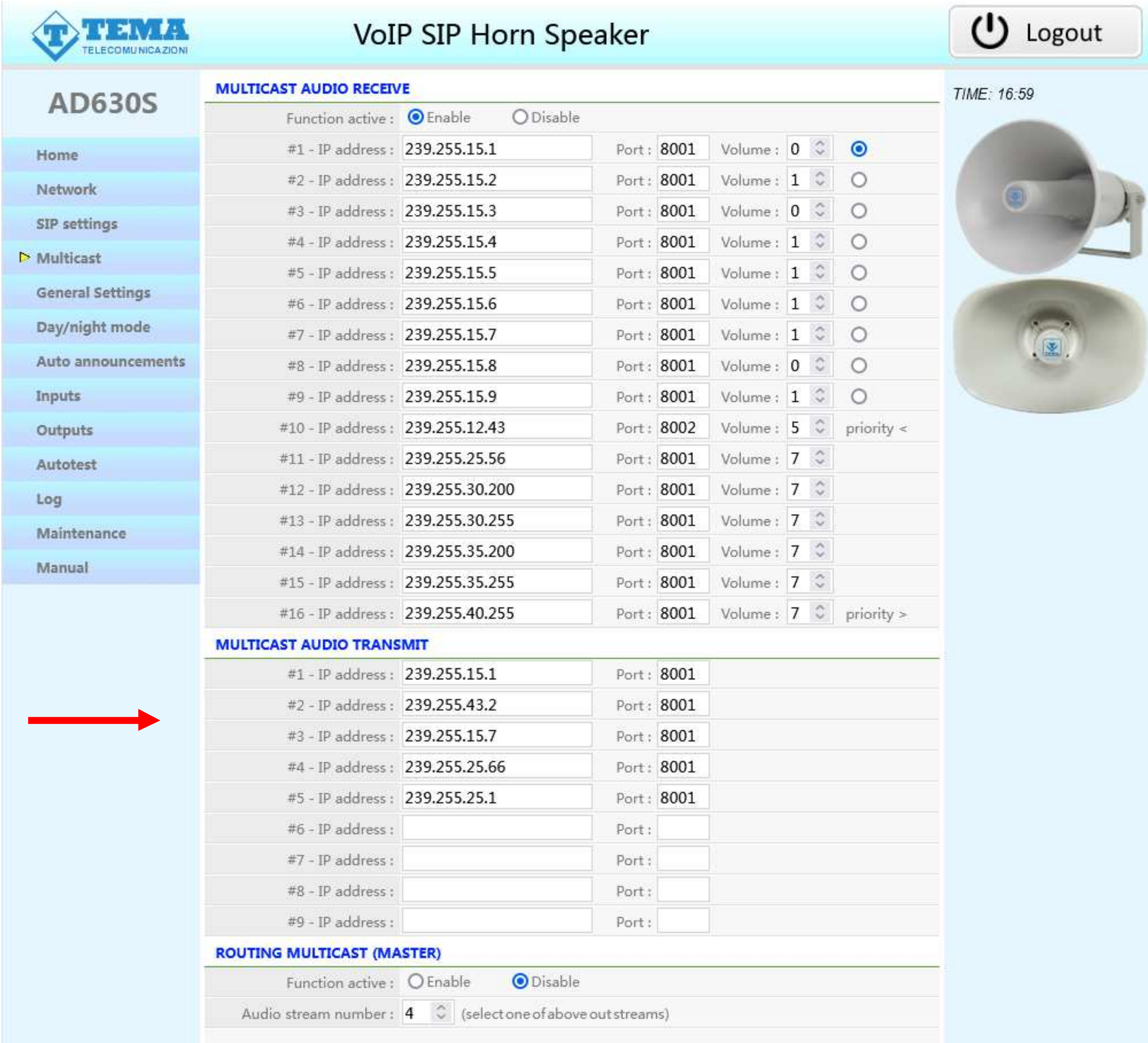
9.4. SIP Parameters

Once the network parameters have been correctly set, the VoIP-SIP connection with the PBX must be configured:

- *SIP server address:* indicate the IP address of the SIP switchboard to which the AD630S must connect.
- *Port:* is the number of the SIP port of the PBX with which to forward incoming calls. Generally it is the 5060 but some stations use another port..
- *Domain:* enter the domain to which to register.inserire
- *Outbound proxy*: some control panels require that the extension number to be called is followed by the proxy address. In most cases this field can be left blank.
- *Local SIP port:* port 5060 is generally used locally as well. It is possible to specify a different one (necessary in some particular routing programs)
- *Expiration*: indicates every how many seconds the device should check the correct registration of the extensions.
- *Main account user / password:* credentials for registering the extension (which obviously must have already been created on the PBX).

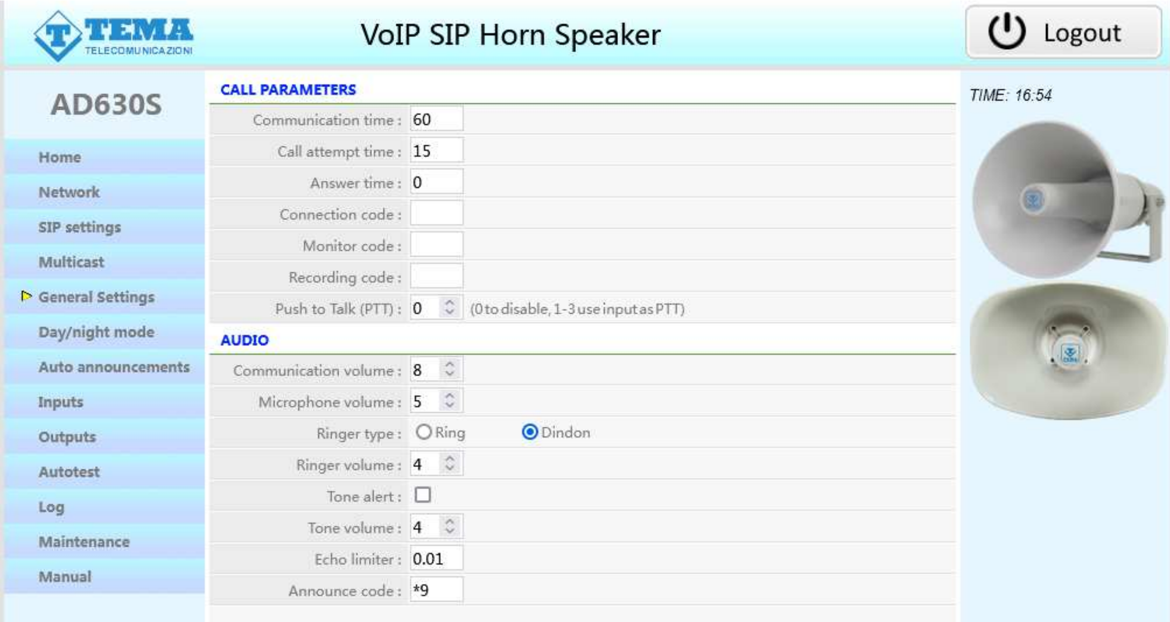
- _Ringer Account User:_ this account is used for the Night Ringer group. In practice, it is possible to program the switchboard to make a group of extensions ring in night mode. You can also put the device in this group, which will play the ring message without answering.
- _Password_: registration password of the second SIP account.

- _Volume_: you can set the volume used to play the Night Ringer ringtone.
- _Ringtone type:_ it is possible to choose between 6 pre-programmed ring tones in the device plus a customizable one (which in the factory coincides with the default one). It is possible to change the customized one from the maintenance web page.
- _Test Ring ON-Test Ring OFF_: buttons that activate the ringer for the purpose of testing the sound emitted
- _Increasing volume_: with this option, the volume will start low and will increase, due to no response over time, until it reaches the maximum volume set in "Ringtone volume".
- _Relay 1-2 activation_: it is possible to choose whether the 2 relays are to be activated during the call and with which mode.
- _Stream replication:_ by enabling this function it is possible to replicate the audio file of the ringtone sound on a programmable Multicast channel in the "Audio stream number" field so that other IP speakers with that Multicast channel programmed to receive can reproduce the same sound. The transmission channel is programmed in the "MULTICAST" page in one of the "MULTICAST AUDIO TRANSMISSION" fields.



9.5. General settings

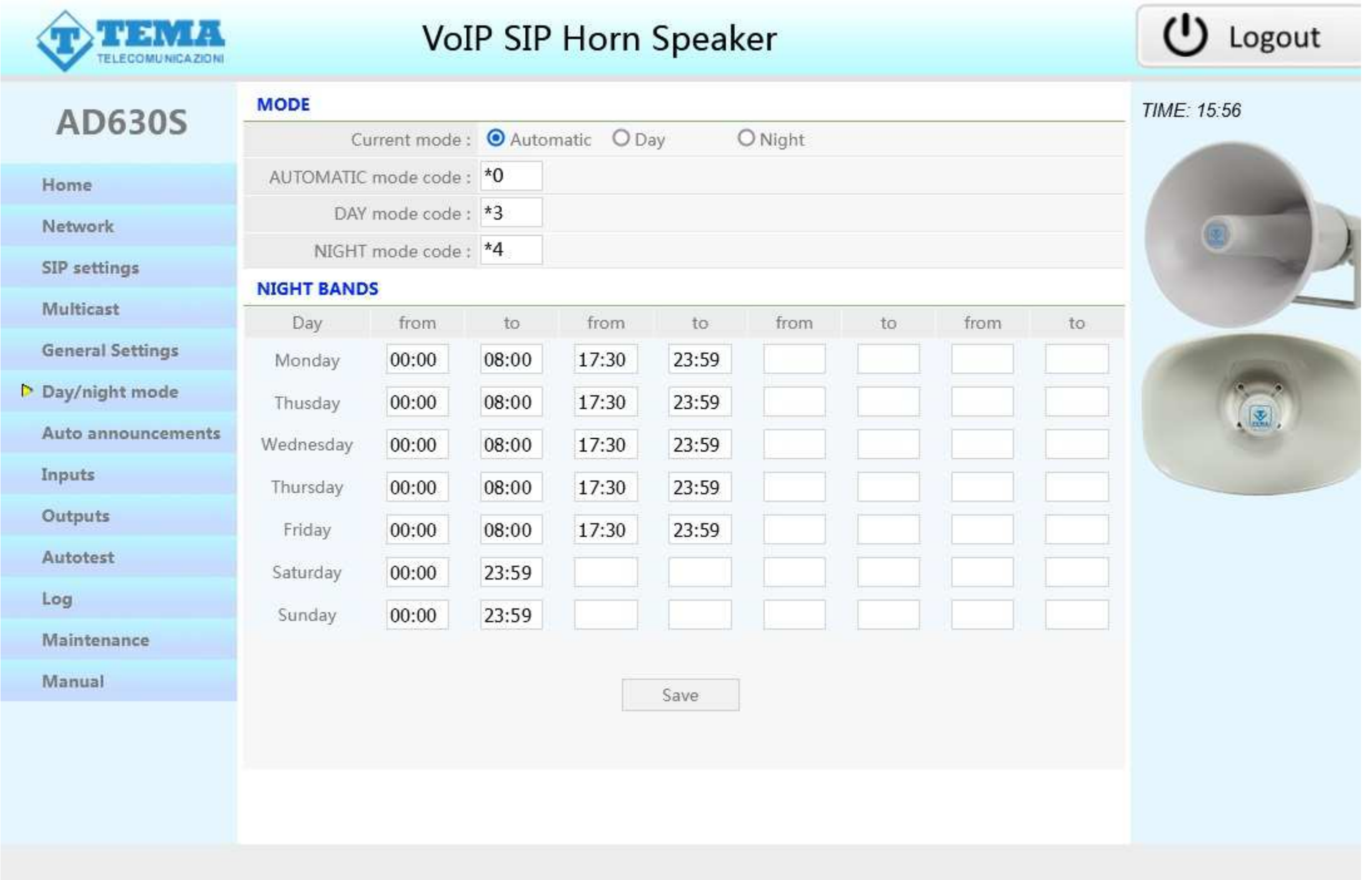
In this section it is possible to program general-purpose parameters, such as call duration, audio volume, tones, ring and so on:



- *<u>Communication duration</u>*: once in connection with the extension, the call is still cut down after the set time.
- *<u>Call attempt duration:</u>* establishes how long the internal call attempt should last, in seconds (if the called number rings but does not answer).
- *<u>Response time:</u>* when the AD630S is called, it answers after the set time. If it is 0, it answers immediately, if 999 it never answers.
- *<u>Connection code:</u>* when the device receives an incoming call, if this field is programmed, the audio is not connected until the user enters the correct code.
- *<u>Monitor code:</u>* as in the previous case, in the case of incoming calls the audio is not activated if this code is not entered, but unlike the previous parameter it allows the connection of the microphone audio only ("monitor" function, it is possible to only listen to the audio near the horn).
- *<u>Registration code:</u>* if this code is present, when the device receives an incoming call, it does not connect the voice line but waits for the user to enter this code. If correct, the system starts recording the caller's message which will be played when the phone call is hung up. It is possible to abort the registration by typing '**'.
- *<u>Push to talk:</u>* allows you to enable the microphone only after pressing a button connected to the programmed input
- *<u>Communication volume:</u>* sets the volume of the audio played to the external user. Possible values range from 0 (very weak) to 9 (very strong).
- *<u>Microphone sensitivity:</u>* sets the sensitivity of the microphone and consequently the volume of the audio reproduced towards the user within the company. Possible values range from 0 (very weak) to 9 (very strong).
- *<u>Ring type</u>*: select the type of ring for incoming calls
- *<u>Ringtone volume:</u>* sets the volume of the ringtone sound emitted by the horn for incoming calls. Possible values range from 0 (silent ringing) to 9 (very loud).
- *<u>Signal tones / Tone volume</u>*: it is possible to give signal tones to the external user (key press, call in progress, end call), set the volume of any signal tones issued. Possible values range from 0 (very weak) to 9 (very strong).
- *<u>Echo Limiter threshold:</u>* this parameter indicates the audio level above which the voice exchange between the caller and the callee can be carried out. Leave the default value.
- *<u>Announcement code:</u>* in communication, if you enter this code followed by a number from 1 to 9, one of the 9 pre-recorded announcement messages is played.

9.6. Set Day/Night mode

It is possible to set the DAY / NIGHT mode manually or automatically by setting weekly time bands. On this page, the codes for changing the modes via remote telephone and the night time bands for the automatic mode are configured:



- *Current mode*: allows you to set the current operating mode.
- *Mode change codes*: enter the change codes to be used remotely (from any internal telephone, call the horn and on answer enter the code corresponding to the selected mode via DTMF). Codes must consist of 2 characters.
- *Night bands*: in the case of automatic operation, it is possible to insert up to 5 NIGHT setting day bands. Outside these ranges the system is in DAY

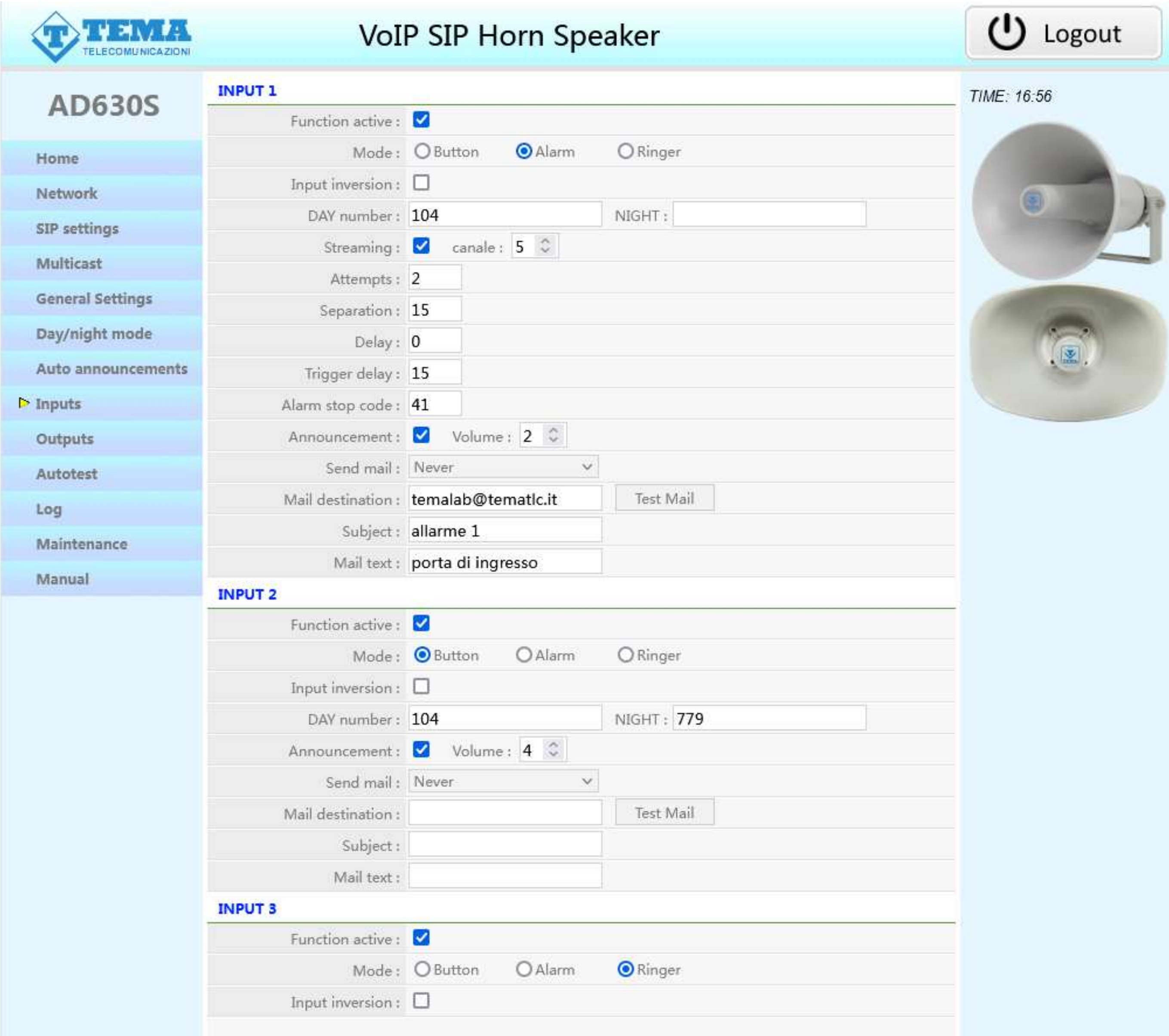
The fields in the table indicate in which time slot the system is in the night. Outside these time slots, the device is in the day. In the example from Monday to Friday the system is in the night until 8 in the morning and then from 17:30 until midnight. Saturday and Sunday it is always at night. At other times it remains in day mode. If the table is left empty, it will never switch to night mode if the function is not desired.

The day / night mode discriminates the numbers to call following the pressure of the call keys or the activation of the alarms.

NOTE: the automatic setting can only work correctly if a time server is programmed in the network configuration web page.

9.7. Inputs

The device has 3 inputs, which can be configured as call buttons, alarm inputs or ringer:



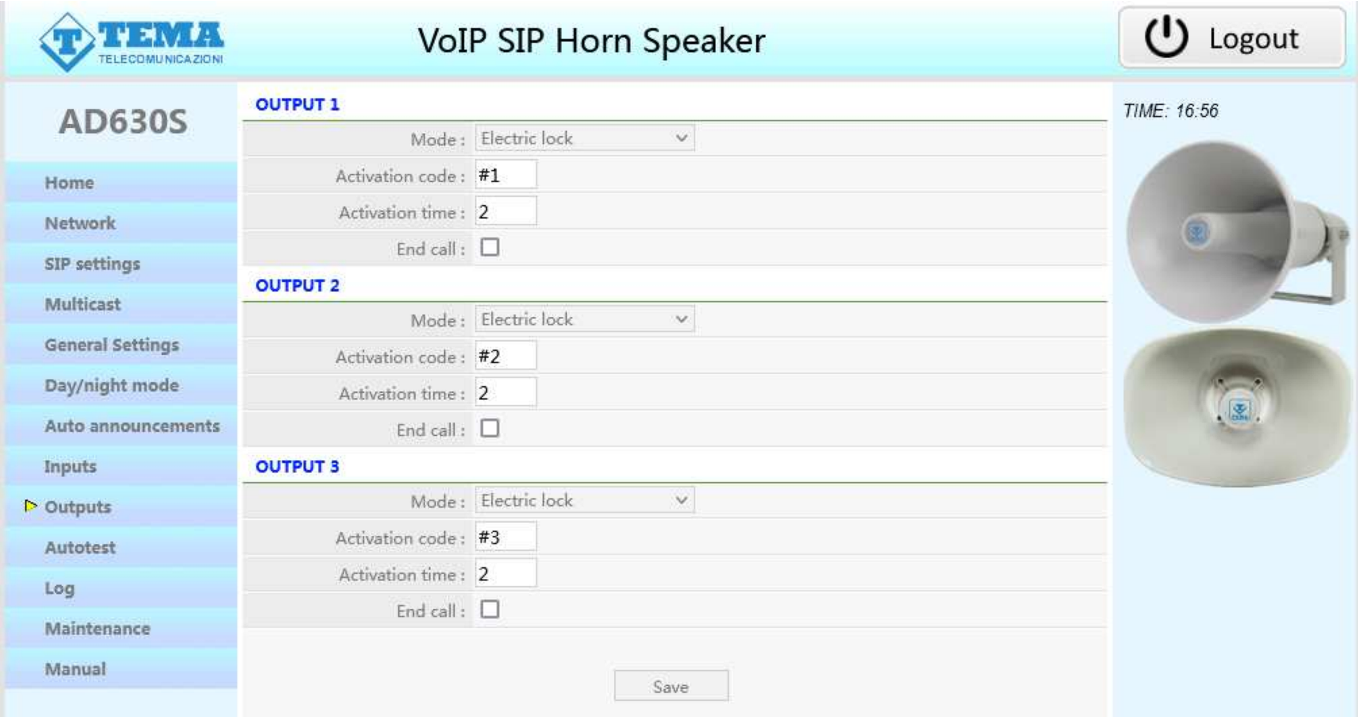
Each input can be configured separately:

- *Active function:* indicates whether to check the input or ignore it.
- *Mode:* in "key" mode, upon activation, the system makes a call and, in the event of an answer, puts the caller and the called party in communication. In "alarm" mode, on the other hand, a one-way call is made and the prerecorded message associated with alarm x is automatically played on response. Finally, in ringer mode, it can act as a repeater: when activated it behaves as if a call came to the ringer account (it would use its programming).
- *Inversion:* indicates whether the activation of the input is to be considered upon closing the contact or upon opening (inversion)
- *Day / night number:* number to call upon activation of the entrance both day and night
- *Stream*: when an alarm is triggered, after having given any warning message locally and before making the telephone call, the alarm message is played to the indicated destination (see the outgoing strem programming)
- *Attempts*: valid in alarm mode. Number of times the system makes a call to report the alarm. Calls are terminated when the silence code is entered. NOTE: if a number to call is not indicated, the alarm message is played only once.

- *Separation:* valid in alarm mode. Separation, in seconds, between one call attempt and the next.
- *Delay:* valid in alarm mode. Activation delay in seconds. It is used to cover any short-term events for which an alarm is not desired.
- *Waiting for rearming:* it is used to prevent the input from reactivating within a certain number of seconds. Think for example of a sensor used to give a message to a passing person.
- *Silence:* valid in alarm mode. Silence code to interrupt the series of calls.
- *Announcement:* upon activation of the entrance, the relevant announcement is issued locally.
- *Sending email:* when the alarm is activated or when the button is pressed, it is possible to send an email with a predefined text to the specified address. The sending credentials are those programmed in the "Network" page.

9.8. Outputs

The standard device is equipped with 2 relay outputs (number 3 is valid only for special versions that require a change to the hardware):



- *Mode:* sets the function to be associated with the output. You can choose between:

  1 – Electric lock: it is activated only after receiving the activation code.
  2 – On call: activated with a conversation in progress until the horn returns to rest
  3 – Pressing the button: pressing any button activates the output for the set time
  4 – Alarm activation: activated for the entire duration of the cycle of any alarm
  5 – Stream: the output is activated when any audio stream is received.

- *Activation code:* relay activation code to be dialed from the internal telephone to the PBX (compulsory 2 characters), valid in electric lock mode.
- *Activation time:* relay activation time
- *Call closure*: by setting this option, the conversation is immediately cut down when the relative relay is activated

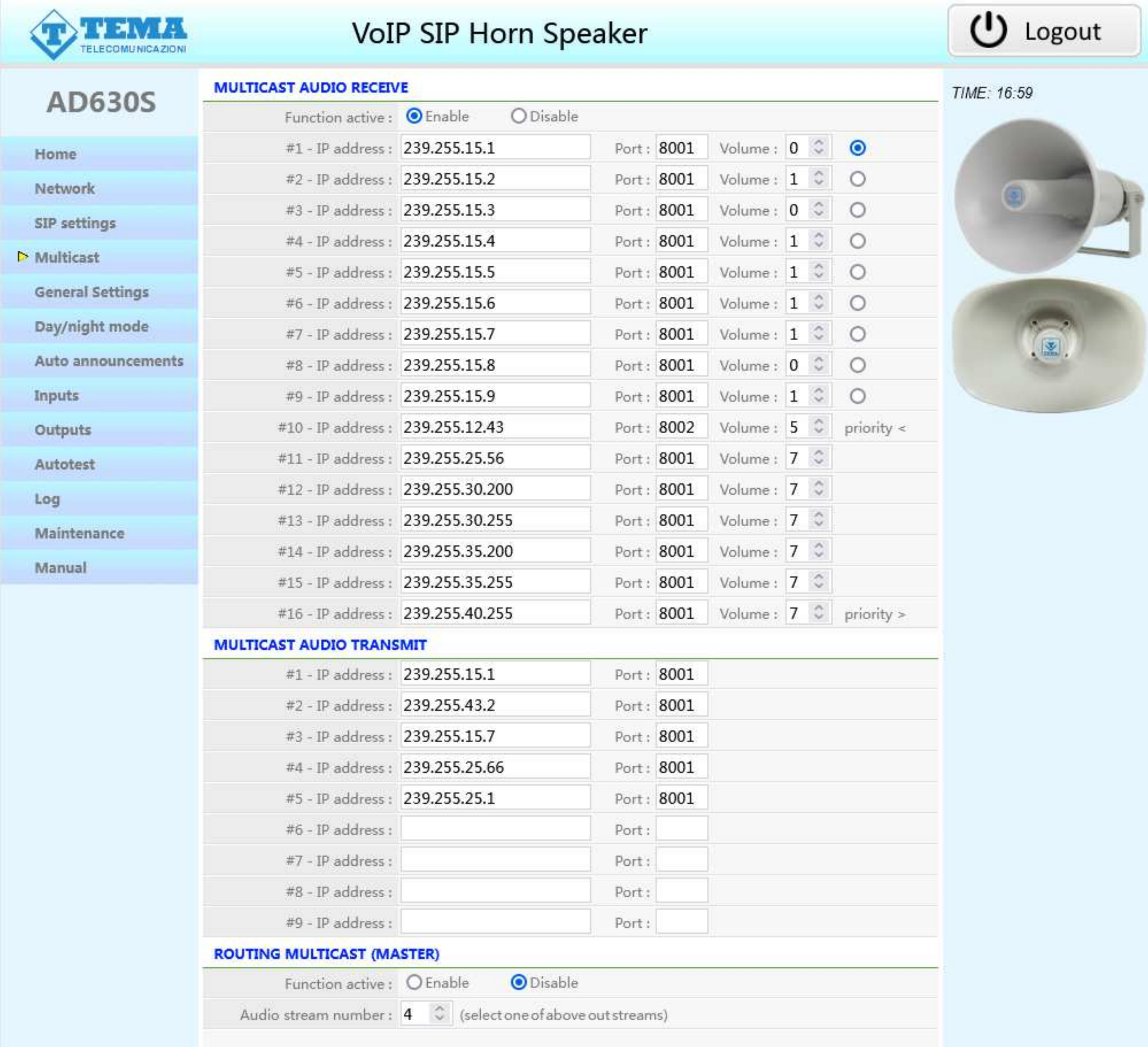
9.9. Maintance

In this page it is possible to change / listen to the pre-recorded messages of “Alarm”, “Announcement” and “Ringtone”. The memory contains default messages for each of the positions, the messages can be customized by the user at will as long as the WAV PCM 8 kHz, 16 bit, mono format is used and that each message does not exceed 60sec in length.

In this same screen it is also possible to save or restore the configuration, change the access password or change the firmware of the device.

9.10. Multicast

By activating this function, the AD630S will listen to the specified address / port, reproducing any received audio stream from the speaker. The function has a lower priority than the normal telephone operation of the system, i.e. outgoing or incoming calls block or prevent the reproduction of the audio stream:



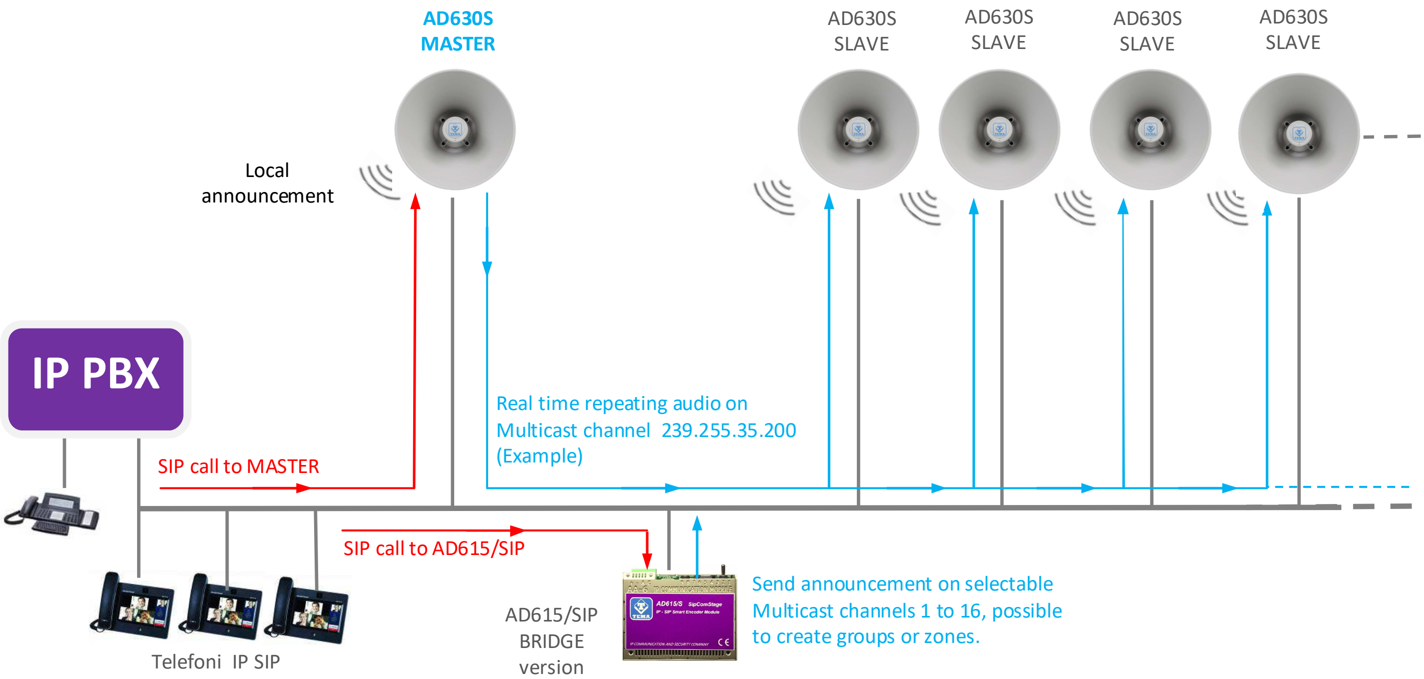
The first 9 Multicast channels are fixed and not linked to priority. You can choose only 1 among them and it allows you to choose, for example, the music you want to listen to if several audio streams are transmitted on the network at the same time.

The last 7, on the other hand, are priority. They can interrupt the audio of the first 9 and in turn be interrupted by a higher priority stream. For example if I am playing a music channel, it will be interrupted by a service announcement.

Each audio stream can have its own playback volume. On the other hand, there are 9 streams in transmission. The purpose of these is to stream pre-recorded announcements (automatic announcements), transmit alarm messages or route the audio received during a call to a destination.

AD630 in Master-Slave configuration for general telephone call on Multicast channels

In the scenario shown, each AD630S can receive the telephone call and broadcast the announcement in its area, without prejudice to all the other programmable functions. The AD630S system configured as "MASTER" receives the telephone call and, in addition to broadcasting the announcement in its area, simultaneously generates a Multicast audio stream on a programmable channel and broadcasts the same announcement on all "SLAVE" loudspeakers enabled to receive the Multicast channel programmed in the “Master” unit.



By inserting an AD615 / SIP device in the scenario it will be possible to further diversify the zones since the AD615 / SIP answer allows you to select from your telephone up to 16 Multicast channels (Zones) where to transmit an announcement through the telephone.

9.11. Automatic announcements

It is possible to program announcements at set times, which can take place on the local speaker or in streaming to other devices.

Daily activation:

it is possible to automate the issuance of a daily announcement (which will take place at the set time) for each day. It will be played for the number of repetitions, with separation between one message and the other given by the number of seconds set. It is also possible to manually activate the message (with the “Activate immediately” button) or stop playback (with the “Stop” button). If you use only the manual activation mode, you do not need to indicate the start time.

The message reproduced is one of the 9 alarm announcement messages found in the “Maintenance” section.

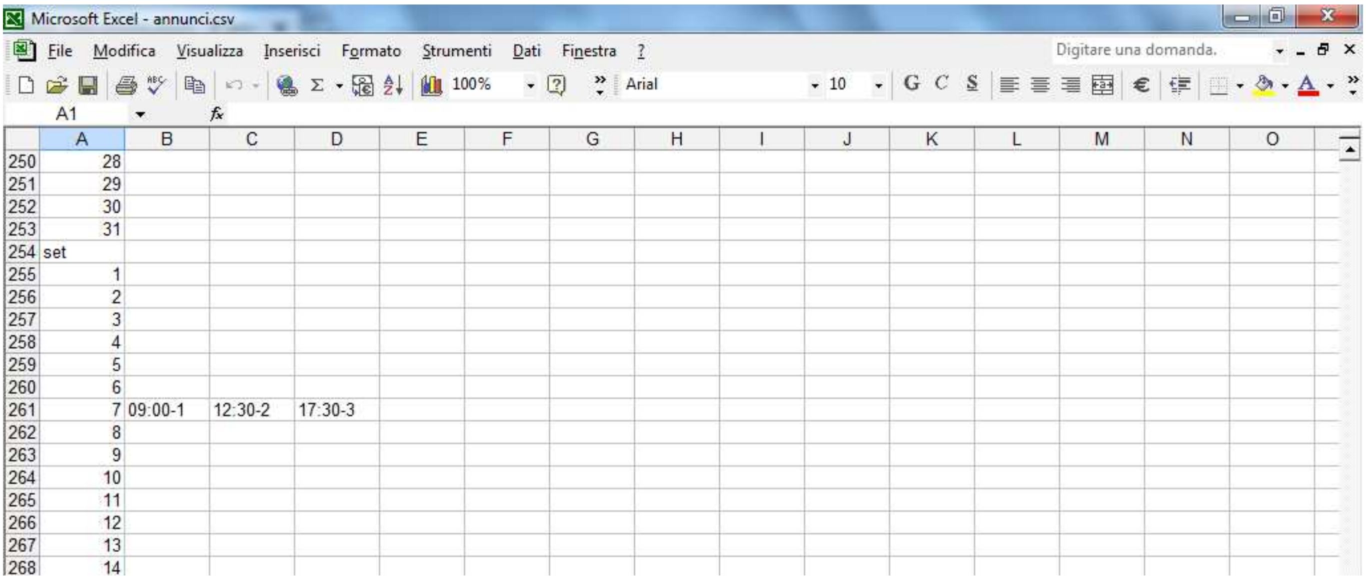
Periodic activation from table:

if this function is activated it is possible to indicate in the table below, for each day of the week, up to five different times, for each of which it is possible to specify one of the nine prerecorded announcement messages.

A number from 1 to 9 must therefore be programmed in the "msg" field (the number indicates the announcement message).

Periodic activation from file:

you can upload a text file in "csv" format (therefore easily editable with Excel ") that contains, for each day of the year, up to 32 times in which to issue the announcement. To compile the file, simply download the default one already present (and empty), modify it and reload it (do not change the words with the months, they are used by the device to index within the document):



	A	B	C	D
250	28			
251	29			
252	30			
253	31			
254	set			
255	1			
256	2			
257	3			
258	4			
259	5			
260	6			
261	7	09:00-1	12:30-2	17:30-3
262	8			
263	9			
264	10			
265	11			
266	12			
267	13			
268	14			

In the example it was indicated that on September 7 (of each year) message 1 must be played at 09:00, message 2 at 12:30 and message 3 at 17:30.

Streaming

Finally, it is possible to play the indicated messages not to the local speaker but in streaming to a multicast address. To do this it is sufficient to insert, after the message number, the letter S followed by the multicast channel number programmed in the “Multicast transmission” mask. For example 1S2 means stream message 1 to the programmed Multicast address 2.

NOTE1:

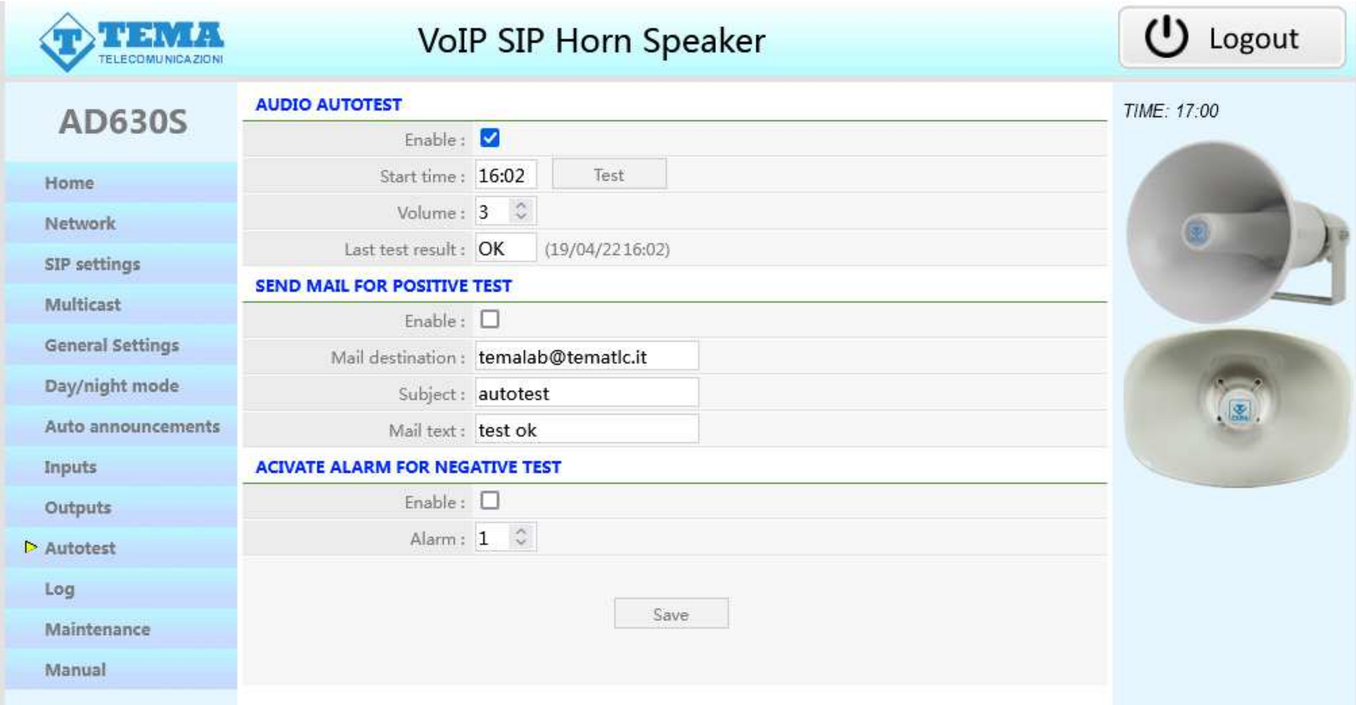
You can set the volume of the message on the local speaker and possibly the activation of relay 1 or 2.

NOTE2:

Since the device is not equipped with its own internal clock, the automatic transmission of messages can only take place if it is synchronized with a time server (which can be local to the network or any on the Internet, see "Network parameters").

9.12. Autotest

To ensure the efficiency of the speaker, it is possible to set the self-test function, to be carried out once a day. In practice, the system automatically emits three acoustic tones on the speaker which are then reread and analyzed through the internal microphone. The system evaluates the quality (or even the absence) of the reread tones and, in the case, behaves as programmed:

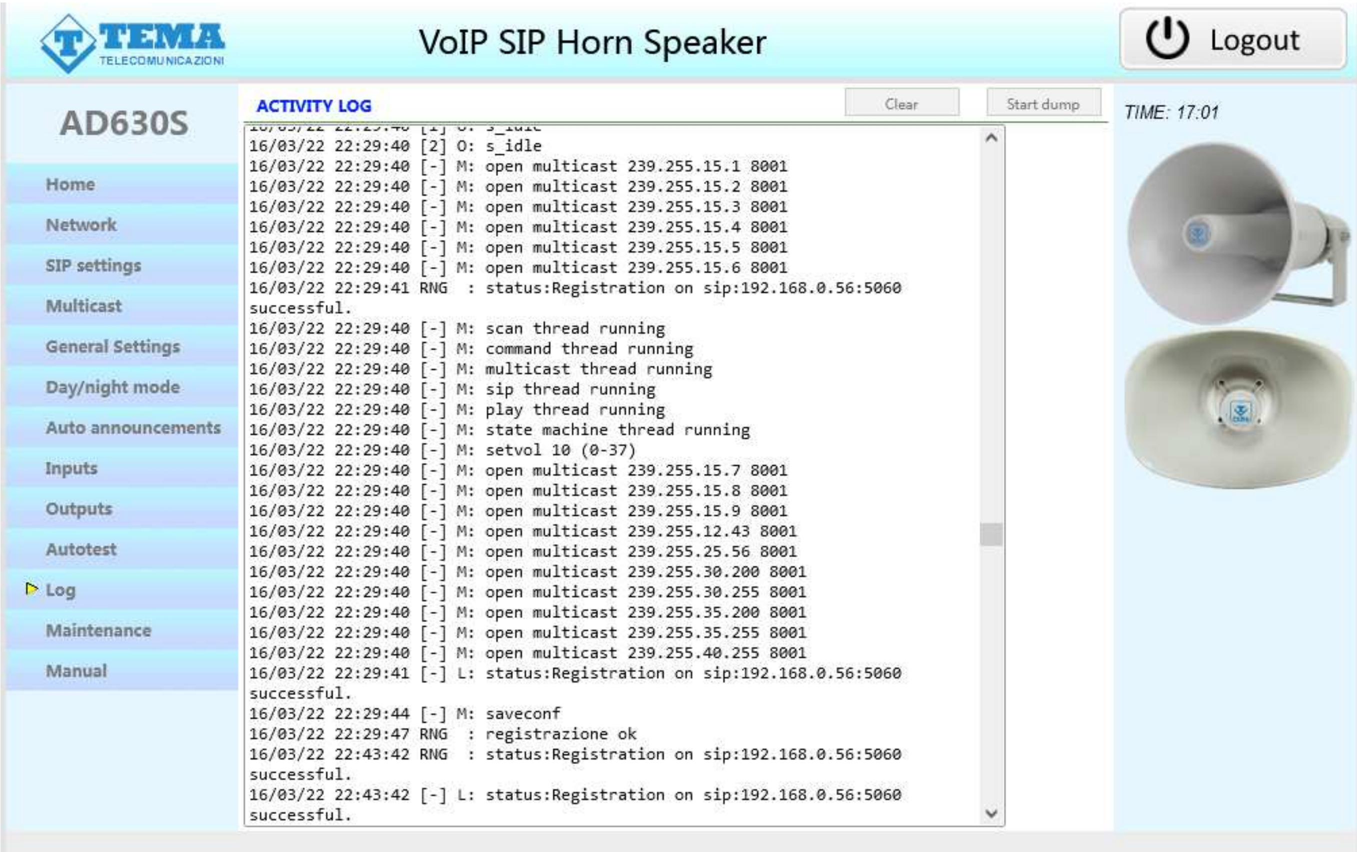


You can decide to send an email with a positive outcome (and in this case you must receive an email certifying the correct functioning of the device every day) or simulate an alarm in the event of a negative outcome (in which case the device will follow the programming of the relative alarm).

The tones are designed not to be annoying but since they are still audible it is advisable to perform the self-test at times not manned by personnel. Furthermore, the volume for the emission of tones must be evaluated on the basis of any environmental noise (if the environment is silent, such as an office, a low volume is sufficient, if it is noisy, such as a mechanical workshop, it is good to choose a sufficiently high volume).

9.13. Diagnostic Log

To identify small configuration problems, it is possible to have a textual diagnostic relating to the activity of the VoIP channel of the device:



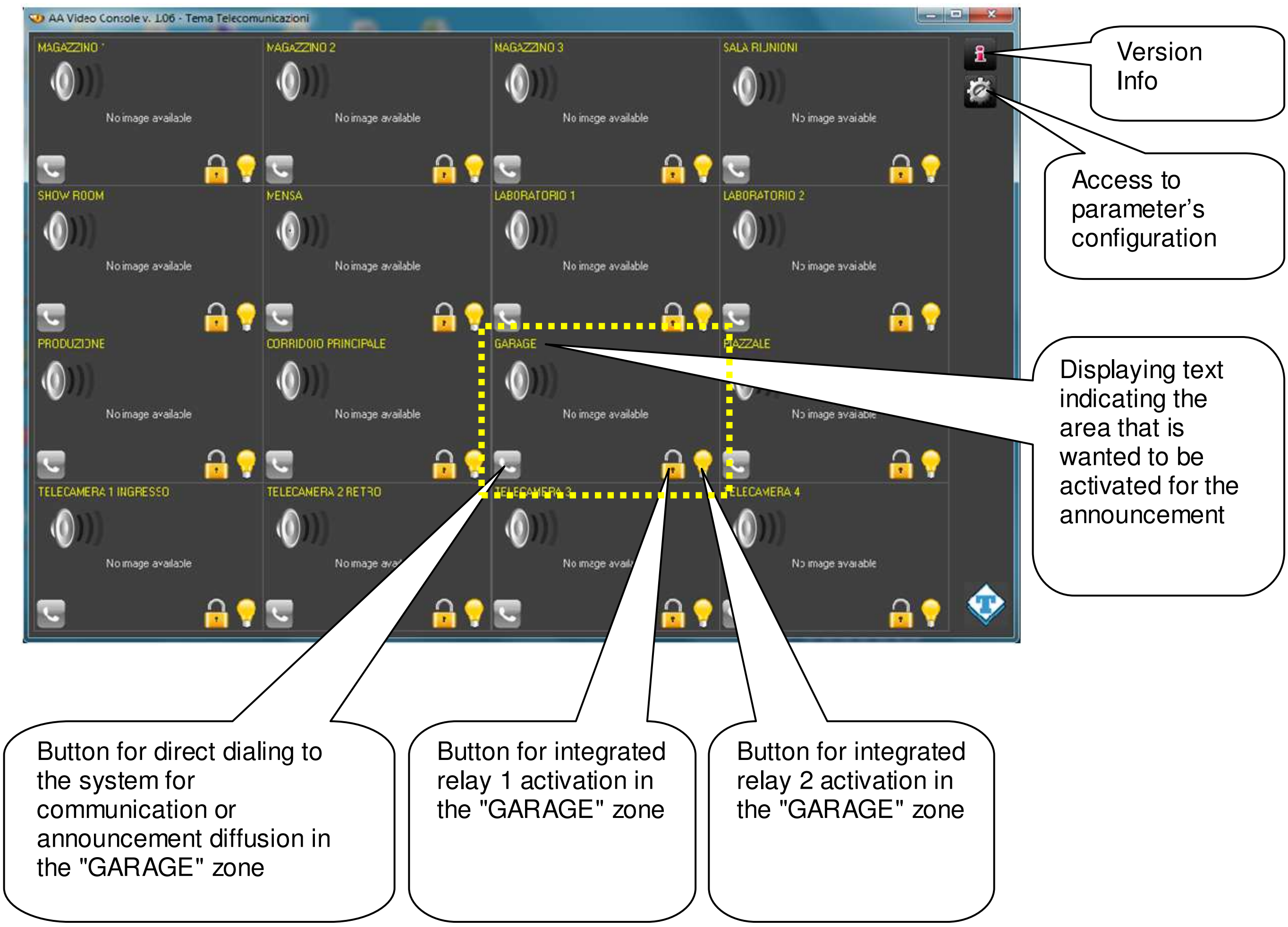
It is also possible to perform a low-level network trace with the “dump” button. A file in pcap format readable with the Wireshark program will be created, freely downloadable from the web.

10. PRESENTATION AND USE OF THE "AA Video Console" SOFTWARE for Windows®

AA-Video Console is an application that comes for free together with the IP audio devices AD600-AD630S and IP Video door systems series AA-500 and AA-600, is used to monitor and operate from a single location with 16 audio devices and/or IP videophones. The software is installed on one or more PCs (with no licensing restrictions) with the operating system from Windows XP® or later. The program allows to see the area framed by the built-in camera in the IP Intercom devices, to communicate and remotely operate the relay of managed terminals in the network, all from the same program window. The PC where AA-Video Console is installed must have a sound card and equipped with speakers and a microphone. It is possible to display up to 16 scenes/devices simultaneously and operating with a terminal at once chosen from those in the window. To install the program identify in the supplied CD-ROM the file "AAVideoConsole/setup.exe".

10.1. Presentation

This is the main program window. In the case of IP AD630S speakers that are not equipped with the camera, the program shows to the operator the IP speakers available to make announcements from PC. Into the network it is possible to have more than one PC running AA-Video Console program. The operator chooses the area where must perform the announcement without having to remember the device number to call (operating mode which is still available) simply by pressing on the handset symbol present in the box of the desired area.



Pressing the handset icon, the program establishes connection with the related terminal and the operator will be able to spread his message by speaking into the microphone connected to the PC.

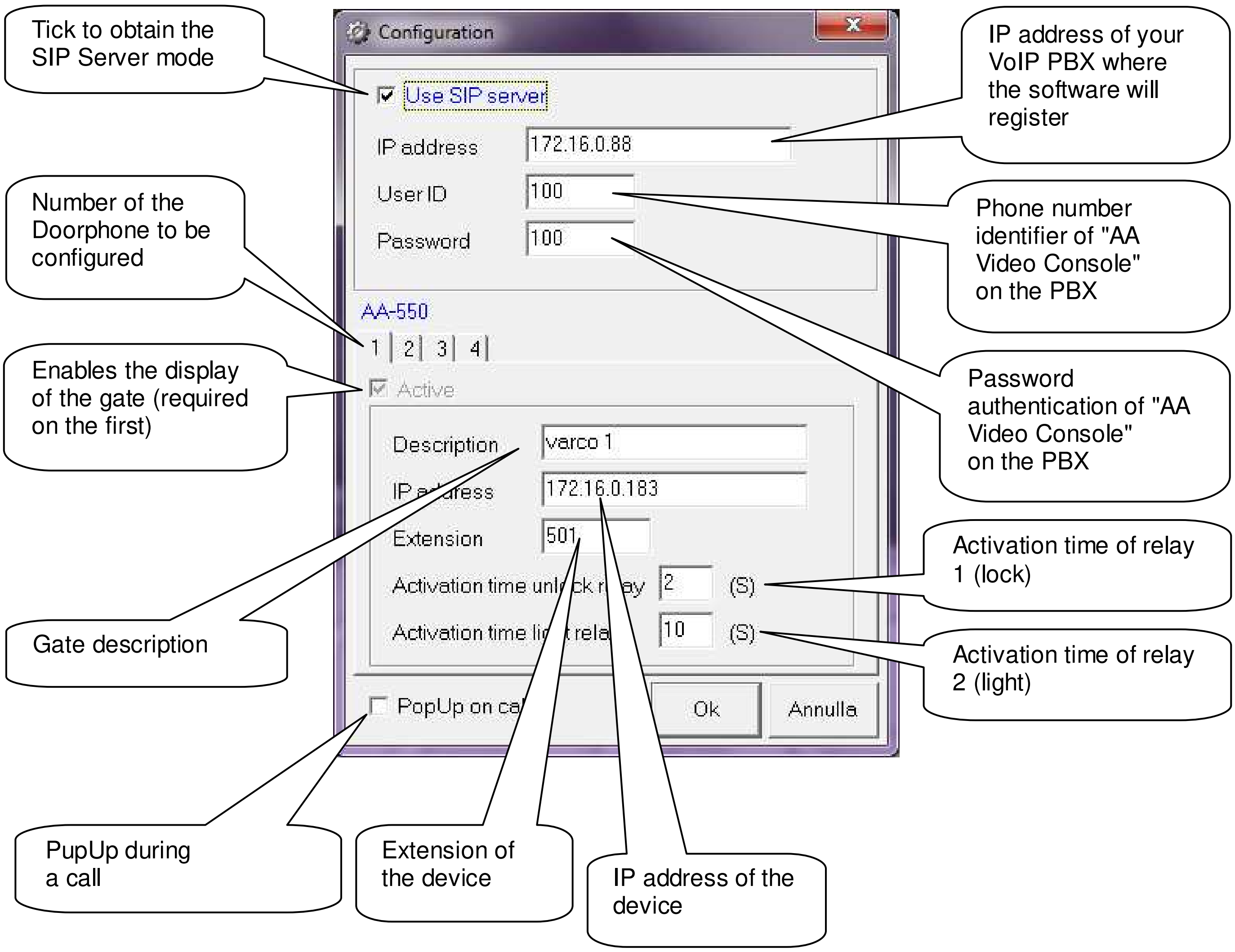
Changing area is easy, just click on the new desired icon and the connection to the previous announcement is terminated and it is established another to the other area for the new announcement.

10.2. Software configuration for SIP Server mode

It is possible that the company owns or installs between the various IP AD630S audio devices also one or more IP Video Intercoms Tema AA-500 and AA-600 series. These can have the camera that checks and shows in window live images from the framing area. Therefore the PA system can incorporate the control of corporate access points with IP video Intercoms whose video streams are displayed in the windows of the AA-Video Console program.

The audio of the conversation with the devices or the voice communication takes place as a normal VoIP communication between two SIP phones. In this case one of the interlocutors will AD630S (or any IP Videophone), the other is the PC with the "AA Video Console" program.

To make sure that the PC running this program can communicate using AD630S with a user, it is necessary to set the number assigned to "AA Video Console" for that device. By clicking on the configuration parameters button of the program, will appear the window shown below. It is possible to configure up to 16 systems. Select the System tab number to be configured in the AA-Video console and for each one will be the relevant parameters. By activating the checkmark "Active" from the second system, into the main page of the program will adapt in amplitude based on the number of active configured systems.



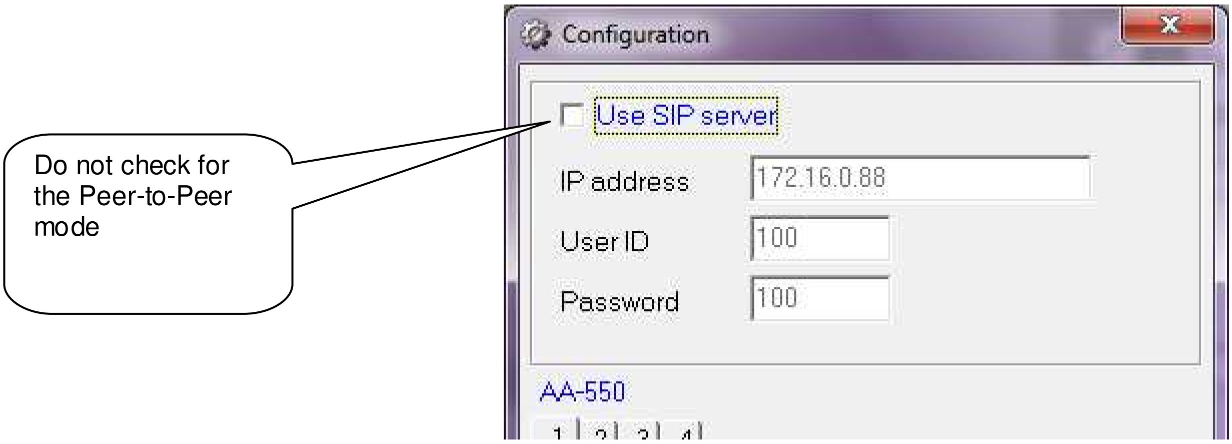
By customizing each description line for the operators will easily and immediately identify the system on the PC screen to be activated to perform the voice announcement from the PC microphone.

10.3. Software configuration for Peer-To-Peer mode

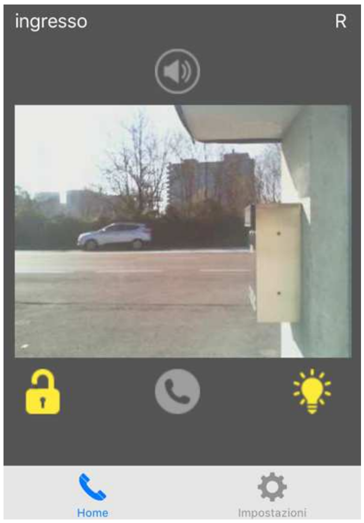
In this configuration mode, the software does not register on a PBX, but makes calls directly to the IP address of the specified device. So the video stream and the audio of the conversation with the device will take place without the intervention of the PBX (the two devices doesn’t need to be registered to any existing SIP Server).

In this configuration, one of the interlocutors will be the IP audio terminal or the video intercom, the other is the PC with the "AA Video Console".

To ensure that a user who is in proximity of the terminal (set up with a call button), can "communicate" through the terminal itself with this program on the PC, it is necessary to of course set the IP address of the PC in the terminal parameters (AD630 or AA-500/600). Doing so, after pressing the button, the terminal will forward the call to the IP address of the PC where the "AA Video Console" software is installed.



11. Introduction to AA-Video Console APP version for operating systems iOS



AA-Video Console is a free app that can be installed on any Apple iPhone® smartphone.

The program allows to see the area captured by the built-in camera in all the IP VoIP SIP TEMA Audio and Intercoms, to operate the device relays and to speak directly with the visitor. The application can be downloaded from the Apple AppStore, can be easily found typing AAVIDECONSOLE in the search box.

It is possible to configure up to 4 terminals and on your iPhone it will be showed a live image by the selected system camera (to switch from one system to the other just scroll on the display): the video streaming is always displayed, even without communication or calls in progress (continuous video monitor).

Offered services

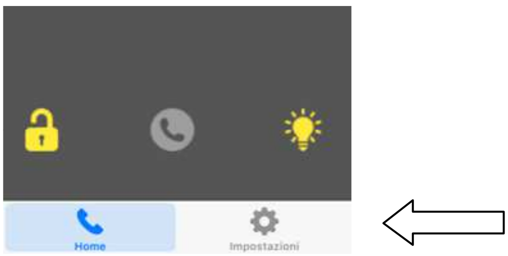
- Sip calls to the network terminals, launching advertisements on IP SIP speakers
- Operation with both SIP Server and Peer-to-Peer mode
- Management of the door relay or step marker light (relay 1 and relay 2) even without necessarily being in communication
- Hands-free audio
- Display of the camera image even when you are not in communication (one Doorphone at a time)
- Up to 4 Doorphones management

11.1. Downloading and installing the app

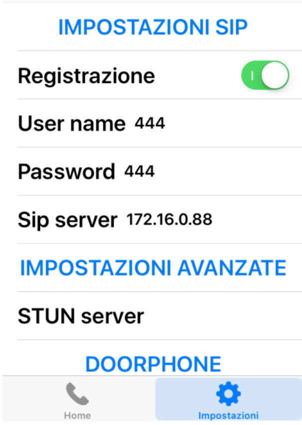
Like all the iPhone Apps, the installation can take place only through the official store. Inside the appstore, simply type AAVIDEOCONSOLE in the search box and confirm the installation:

Configuration

Both the general configuration and the individual Doorphone configuration is done in the "Settings" menu:



You will see the following screen:
:



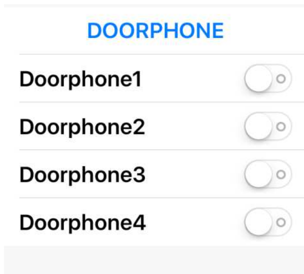
The first part, "SIP SETTINGS", allows to register your device to a PBX or a SIP provider (necessary if you want to reach the Doorphone through its extension number). Marking "Registration" you will have access to the account setup registration (user name, password, etc.). If into the network a SIP PBX is not installed and want to reach the Doorphone simply by its IP address, leave the check unmarked.

In the "ADVANCED SETTINGS" it is possible to configure any STUN Server (external to the private network) that allows the application to reach the Doorphone through a firewall.

NOTE: to access the Doorphone from outside the corporate network (ie from the Internet) it is necessary to program your router / firewall to allow incoming connections to an internal device. For this feature contact your network administrator (each router / firewall has its own programming that it is not possible to generalize in this manual).

Configuring Tema IP SIP terminals

It is possible to control with the application up to 4 systems simultaneously.
Also in the settings page is the menu to configure them:



Activating the slider of each Doorphone (or IP audio system), the fields to be configured will appear:

The description appears in the displayed image and is useful to identify the selected Doorphone from time to time.

In the address field you must enter the IP address of the Doorphone. It is used to show the camera or to make a call in Peer-to-Peer.

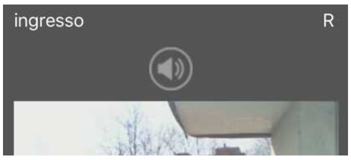
The port number is used internally by the program for the communications, must be 8081.

The extension number is the number that must be dialed when from the main screen is selected the call key. Obviously it makes sense only in SIP registered to the PBX Server. In Peer-to-Peer mode this field cannot be filled.

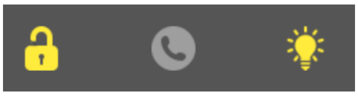
Finally in the field SN you must enter the serial number of the device. For safety reasons, it is needed to enable the opening keys for relay 1 and relay 2).

11.2. Using the application

Once the application is activated, it is possible to check the registration status in the "Home" screen. If you see an "R" in the upper right it means that the application is registered to the SIP PBX.



In normal operation, it is displayed the image of the first Doorphone (as mentioned above to switch from one to another simply scroll the screen). The keys below the displayed image will have impact on the selected terminal:



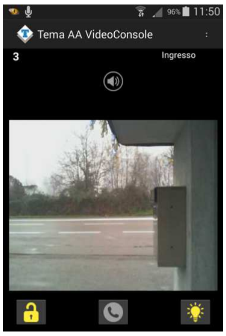
The first button activates the door relay (number 1). The center button makes the call to the Doorphone (or closes if it is in progress). The third button activates the relay normally connected to the light step marker (the number 2).

If the program is put into the background (using the "Home" or "Back" button) it is still active and ready to receive any incoming call (in which case it automatically puts itself in the foreground and is displayed the image of the terminal that has generated the call).

Exit from the App

Proceed as for any other installed iPhone APP.

12. Introduction to AA-Video Console APP version for ANDROID systems



Is a free app that can be installed on any Android® smartphone. The program allows to see the area captured by the built-in camera in all the IP VoIP SIP TEMA Doorphones, to operate the device relays and to speak directly with the visitor. The application is on the supplied CD together with the Windows® version (in apk format). It is possible to configure an unlimited number of terminals and is presented on your Smartphone a live image at that time by the selected system camera (to switch from one system to just scroll on the display): the video streaming is always displayed, even without communication or calls in progress (continuous video monitor).

Offered services

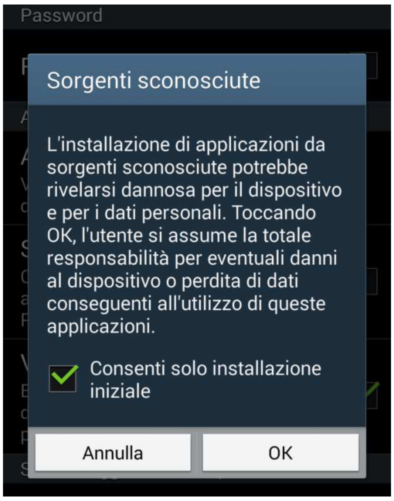
Are the same as the iOS version (see above).

12.1. Downloading and installing the App

Like all the Android Apps, it is possible to install the app without passing from the store, simply by copying and installing the apk provided in the installation disk. To transfer the file to your device it is possible to follow two paths:

Method 1: send an email to the smartphone with attached aavideconsole.apk file (on the CD). Once the email is opened on your smartphone and select the attachment, the system will ask whether you want install the application (which will happen in a few seconds). Probably the operating system will alert the installation from an unknown source:
:

You must select the "Settings" button and allow installation:

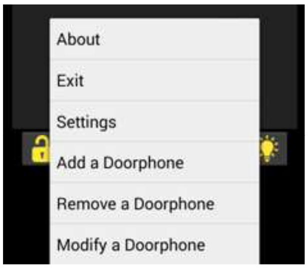


Method 2: copy aavideconsole.apk directly on the device connected to a computer via the USB cable normally supplied with the phone (from "My Computer" the phone is seen like a normal pen drive).

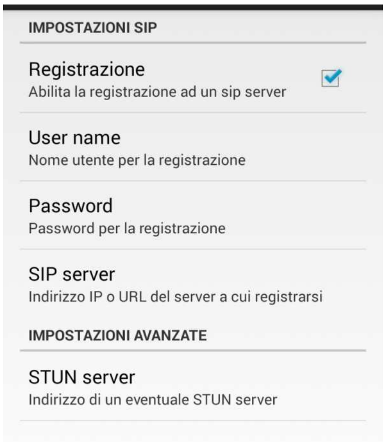
After copying the file you just select it on the phone and confirm the installation (in this case it will also be necessary to authorize the installation from unknown sources).

Configuration

First, from the application menu, the general configuration must be done:



Then click "Settings":



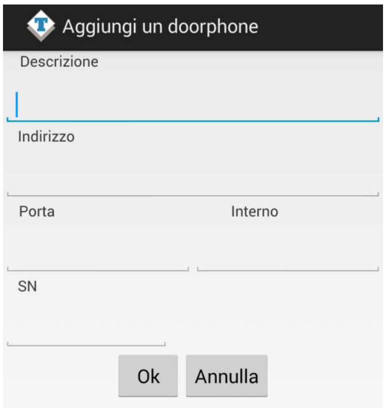
The first part, "SIP SETTINGS", allows to register your device to a PBX or a SIP provider (necessary if you want to reach the device through its extension number). Marking "Registration" you will have access to the account setup registration (user name, password, etc.).

If into the network a SIP PBX is not installed and want to reach the device simply by its IP address, leave the check unmarked.

In the "ADVANCED SETTINGS" it is possible to configure any STUN Server (external to the private network) that allows the application to reach the device through a firewall.

NOTE: to access the device from outside the corporate network (ie from the Internet) it is necessary to program your router / firewall to allow incoming connections to an internal device. For this feature contact your network administrator (each router / firewall has its own programming that it is not possible to generalize in this manual).

Adding Tema IP SIP Terminals



It is possible to add in configuration as many terminals you want. Into the main screen of course it is always presented one at a time.

To switch from one system to the other just slide your finger horizontally on the application display. By choosing "Add a Doorphone" you will see the following screen:

The description appears in the displayed image and is useful to identify the selected terminal from time to time.

In the address field you must enter the IP address of the terminal. It is used to show the camera or to make a call in Peer-to-Peer.

The port number is used internally by the program for the communications. If left blank, it will be automatically filled with the default (8081).

The extension number is the number that must be dialed when from the main screen is selected the call key. Obviously it makes sense only in SIP registered to the PBX Server. In Peer-to-Peer mode this field cannot be filled.

Finally in the field SN you must enter the serial number of the device. For safety reasons, it is needed to enable the opening keys for relay 1 and relay 2).

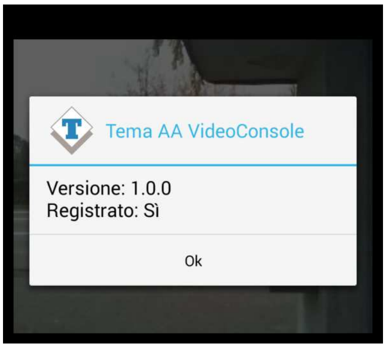
Edit / delete terminals

This feature of course allows to change the configuration of a terminal or to delete an inserted one. Simply select from the list the one that you want to edit or delete:

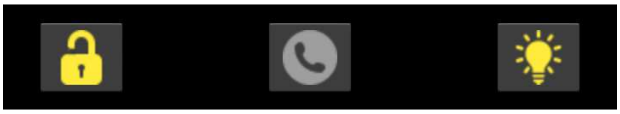
Any changes must be confirmed with the "OK" button. The button "Cancel" or "Back" does not save, useful in case of mistakes.

12.2. Using the application

Once that the application was activated, it is possible to check into the "Info" screen, in addition to the program version, the registration status:



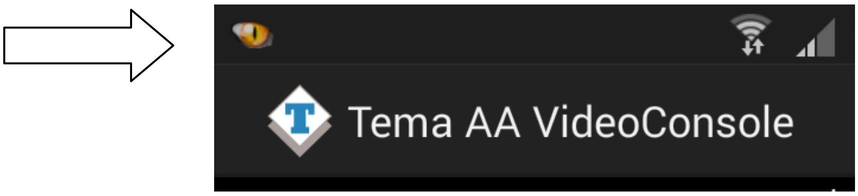
In normal operation, it is displayed the image of the first Doorphone (as mentioned above to switch from one to another simply scroll the screen). The keys below the displayed image will have impact on the selected Doorphone:



The first button activates the door relay (number 1). The center button makes the call to the Doorphone (or closes if it is in progress). The third button activates the relay normally connected to the light step marker (the number 2).

If the program is put into the background (using the "Home" or "Back" button) it is still active and ready to receive any incoming call (in which case it automatically puts itself in the foreground and is displayed the image of the Doorphone that has generated the call).

On the taskbar, the program icon will appear, indicating that it is running in the background:



To restore the application in the foreground is therefore sufficient to select it from the toolbar:

Exit from the App

To end the application, as seen, do not use the "Back" button (like most of the app). To close the program, the "Exit" button must explicitly be selected from the main menu:

13. APPENDIXES

13.1. Appendix 1: Other examples of Multicast Audio Streaming applications

As described above, in a LAN network the Multicast term refers to the possibility to distribute informations like data, audio or video, to a group of IP terminals of the Ethernet network. For multicast are used Class D addresses that range from 224.x.x.x up to 239.x.x.x.

The AD630S systems, in addition to the classic functions previously described in this manual, are able to receive audio in Multicast in the specific IP channel (up to 5 channels) and play it on your speaker. Audio can be generated by a PC software application (for example the free VLC application), from a SIP IP phone or from a TEMA intercom of AA-500 IP series.

It is in fact possible to program each button of the AA-500 IP system for immediately audio sent picked up by his microphone, the received audio stream is played without the intervention of any operator, this function is useful for playing messages and information or emergency announcements (also called "Paging" mode).

SIP phone that has programmable function keys for Multicast calls can be used. Audio streaming using multicast is unidirectional, in the sense that the audio stream is sent from the source (such as a phone) to the destination (AD630S Horns) but not vice versa.

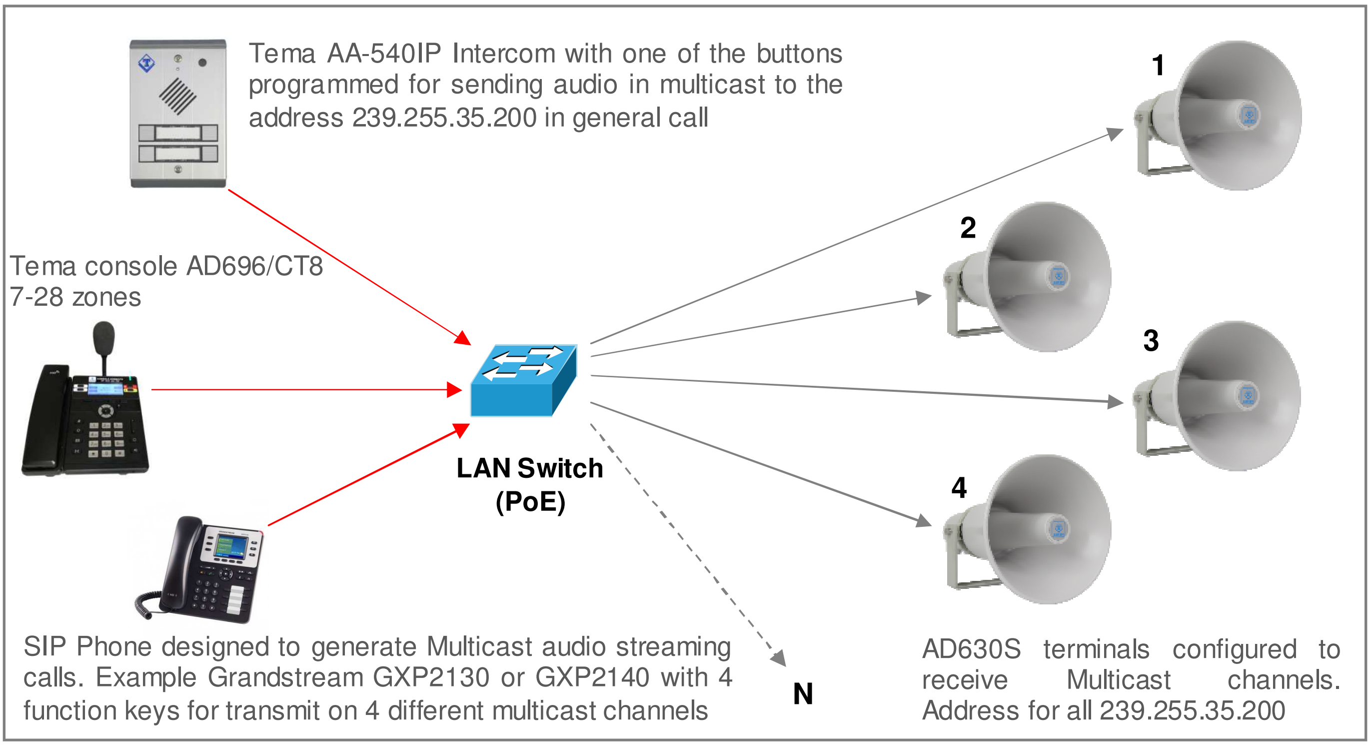
The audio stream can be simultaneously sent to multiple terminals that all have the same IP Multicast address, or to distinct terminals (each has its own IP address).

The audio supported by this mode must be in G.711 format (aLaw or µLaw).

Example of general call in Multicast
The diagram below shows a system where there are 4 AD630 devices connected to the network switch. From a preset telephone it is possible to make a multicast call and launch a voice announcement to the address 239.255.12.42. All the systems are configured with the same multicast address, and then all will reproduce the outgoing audio message.

This function does not affect the normal operation of the AD630 device since the multicast audio is received in a separate channel. At the end of the Multicast announcement, the normal functions are restored.

The use of a PoE (Power over Ethernet) Switch allows to feed the AD630 systems on the same network UTP cable, in this case external power supplies are not needed because the AD630 series systems are compatible with the PoE (Power over Ethernet) standard. **Warning: with PoE power supply the maximum power must be reduced.**



Sending Multicast ads to groups (zones) of IP Speakers

In the plant of the diagram are present 5 AD63x systems configured into three different groups.

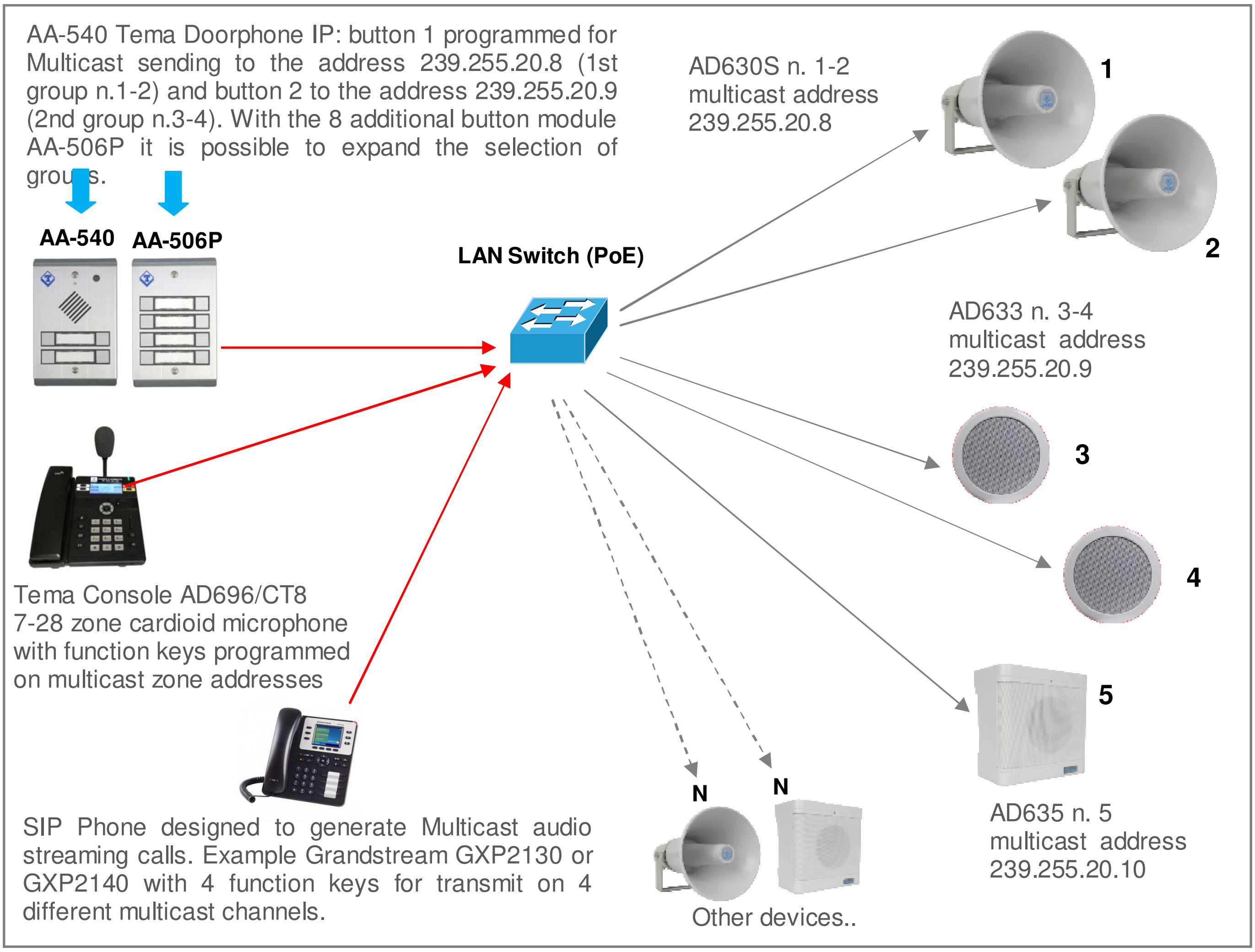
Group 1:	AD630S N. 1-2	IP 30W Horn	with IP address 239.255.20.8
Group 2:	AD633 N. 3-4	Ceiling mount IP speaker	with IP address 239.255.20.9
Group 3:	AD635 N. 5	Wall box speaker	with IP address 239.255.20.10

In SIP telephone, suitable for multicast services, are programmed 3 function keys with the IP address of the groups (Zones) configured so that each function button can send a multicast announcement on associated group.

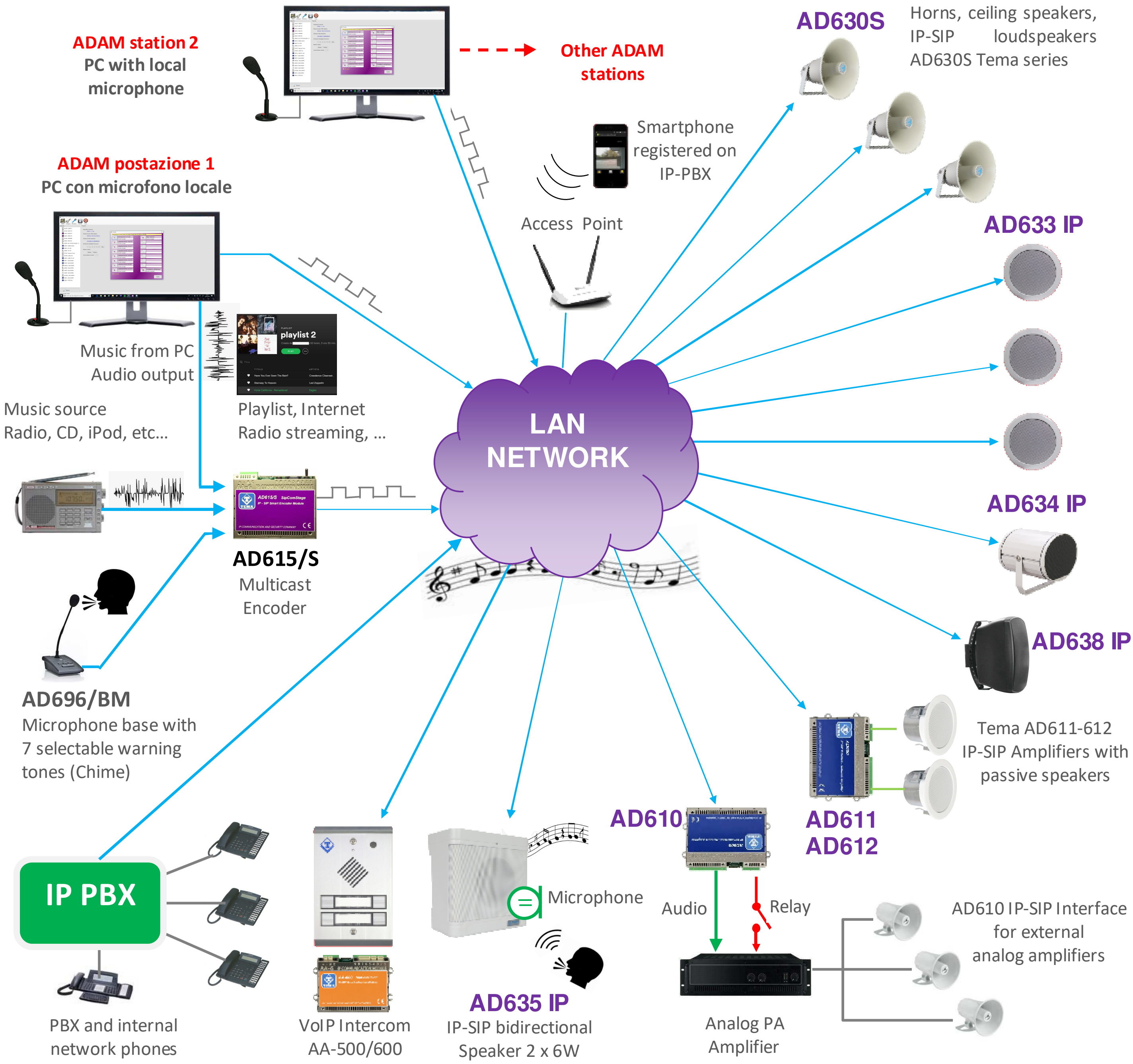
It is also possible to use the Tema AA-500 Intercom to generate calls on Multicast channels by appropriately programming the system keys: holding a key the microphone audio is sent to the programmed Multicast IP address. For further needs of groups (zones) it is possible to add a buttons expansion module in the case of AA-540 or a expansion console of the function keys (DDS) in the case of using a SIP telephone.

Each group can have an unlimited number of AD63x devices according to the sound needs in the served area. The number of groups is expandable up to 10. By request the groups can be further expanded.

The 5 multicast channels in AD63x systems have configurable priority levels, so it is possible to reserve the highest priority channel with a dedicated IP address in order to reserve it for the general call on all AD63x or to create combinations of Zones for ads areas.



13.2. Appendix 2: Scenario of a complete system with Tema ADAM management software



Base Services (Without IP-PBX)

- Different Multicast music according to zone preferences
- Microphone alert messages by zone or all zones
- Messages from IP-SIP TEMA AA-500/600 series Intercom
- Multicast messages from a SIP phone with function keys
- SIP calls in P2P (Peer to Peer)

Additional services in the presence of an IP-PBX

- Bidirectional telephone call on each SIP loudspeaker
- Telephone call from smartphones registered on the IP-PBX
- Messages from smartphone with iOS/Android APP
- Messages from Softphone (Softeclient)
- Night call ringtone repetition (Night Ringer)

It is possible to insert multiple AD615/S encoders each of which transmits a musical sound stream on a specific Multicast channel, in this way the speakers can be tuned to different musical contents according to the needs and preferences of the environments to be sounded. The AD610 decoder allows to interface a traditional amplification system to the LAN network functionality, while the AD611-612 IP amplifiers enable the driving of loudspeakers and passive horns of the traditional system.

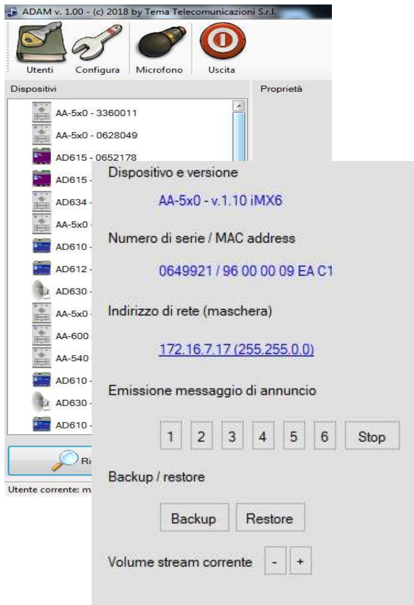
There are no limits in the number of ADAM software that can be installed in the network PC, with licenses for 3-16-32-64-128-256 manageable zones.

Download the documents and the Application Notes of the Audio over IP systems from our website
http://www.tematlc.it/audio-ip.asp

Performance and services of ADAM Audio Domain & Access Management software

TEMA Devices search on the network

This function activates the search for all TEMA IP Audio devices in the local network. The devices are listed with the respective serial number or, at the user's choice, with a description programmed by the user himself in the device, for example the planimetric position can be indicated (Warehouse 1, Warehouse 2, Meeting room, etc.)



Display of main device parameters

With a simple click on the chosen device the main informations are shown: model, firmware version, serial number, MAC address, IP address.

Audio volume adjusting

On the selected device, it is possible to increase or decrease the output audio volume.

Messages playing

In each of the Tema IP audio terminals, it is possible to load up to 6 pre-recorded audio files (Messages, Sounds, Music) and launch them manually from the centralized location with the ADAM software.

Configuration backup and restore

They allow to save and restore the configuration of the device for security reasons

Announcements from local microphone

With this function it is possible with a simple click to send an announcement from the local microphone connected to the PC to a specific Multicast channel with priority over the music channels, in this way any background music is interrupted in the remote devices and the microphone audio will be sent. At the end of the announcement, the music will be restored. It is also possible to send a **warning tone (Chime)** before starting the announcement. The type of warning tone can be selected in the configuration mask among several available.

Sending of pre-recorded audio files

As an alternative to a microphone announcement, it is possible to select an audio file from a PC folder and send it in streaming on a Multicast channel. It is possible to choose both the Multicast channel on which to send announcements or audio files and the sound quality of the stream.

Console Function

With the "Console" function of ADAM launching an announcement from the local microphone to a specific area is an operation that requires only one click on the button of the desired zone, it is possible to start talking into the microphone and end with another click. It is also possible to send a **warning tone (Chime)** before starting the announcement. The type of warning tone can be selected in the configuration mask among several available.

Announcements on multiple zones at the same time

With the same simplicity it is possible to select the zones and start talking in the microphone, another click ends the operation.

Sending of pre-recorded audio files

It may be useful to have pre-recorded audio files to be sent to the IP speakers of the zones at the desired times. Simply select and load the desired audio file from a PC folder into ADAM and activate the sending with just one click.

Scheduling of pre-recorded audio files sending

It is possible to program the automatic sending of messages at pre-established times with daily, weekly, monthly and annual planning.

Generation of 2nd Multicast Stream Audio (Useful for background Music)

Adam is able to capture in digital format any audio content played at that moment on the PC capturing it from its sound card, in this way it is possible to activate a second Multicast Stream that can be sent to a specific IP address normally used by receivers for background music.

For example, if the PC is playing a radio or TV channel streamed from the internet, the content will be sent on the established Multicast channel and played by all the Tema AD600 series decoders in the LAN network enabled for the service.

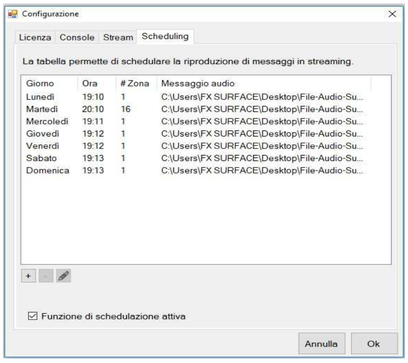
With ADAM, the PC also becomes a multimedia station for the transmission of music and media content on the LAN.

Internet Radio and TV streaming, Youtube, Spotify, iTunes, Playlist, CD players, USB, local Smart Cards, etc.

Sending of pre-recorded audio files at scheduled times or days

With this function it is possible to program the sending of pre-recorded audio files to the desired times and possibly repeat them on the preset days. It is also possible to schedule the announcement of announcements at pre-established intervals, for example, as a promotion program produced in supermarkets, the notice of term lectures in a school, a signal for the end of working hours, etc.

There are no limits in the number of programmable audio files available online in the programming mask of the ADAM software.



Sending of pre-recorded audio files to the microphone channel

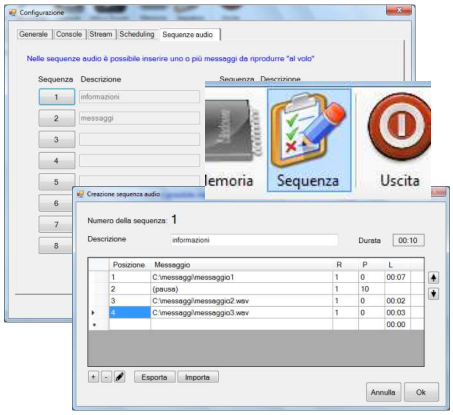
As an alternative to a microphone announcement, it is possible to select an audio file from a PC folder and stream it on a Multicast channel.

Immediate sending of audio file sequences

With this function it is possible to program the sending of audio files which can be consecutive timed messages or music files such as Play List. The programmed sequences, as well as manually, can be recalled and also launched by the "Scheduling" program at set days and times.

Within a sequence the individual files can be moved from position to position freely using position arrows.

There are no limits in the number of audio files programmable within a specific sequence. Thanks to the "Import" and "Export" functions the programmed sequences can be archived, recalled or managed by different computers with ADAM software installed.



Programmable memories of pre-recorded audio files for instant sending

The function allows to record messages through the microphone, store them on disk and recall them for immediate sending to a specific area or to all areas (general sending). 16 memory areas are available in which audio contents (messages but also music) can be stored with a duration of each single file up to 30 minutes. Sending can be interrupted at any time.

The use of this feature is particularly useful in very large environments with accentuated reverberations, where it requires a considerable power of the amplifiers with many loudspeakers and it is not possible to avoid microphone disturbances, except with expensive control equipment: in this way it records the message is sent deferred, when the microphone is no longer active. Or to send repetitive standard messages in different languages.

Remote activation of the relays of all TEMA terminals in the LAN network

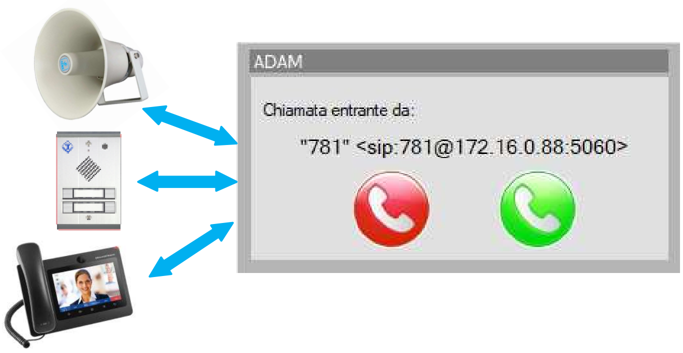
From the ADAM workstation installed in a PC it is possible to select each individual Tema AA-5xx or AD6xx terminal in the LAN network and operate the relays of which the device is equipped with a simple click on the position of the respective relay. In the same way the relay can be deactivated regardless of the set timing.

Integration in ADAM of the "VideoConsole" software

By installing the "VideoConsole" software on the PC where ADAM resides, the relative activation button will appear on the toolbar.

VoIP SIP calls to and from the terminals

The function allows making calls to all AD600 and AA-500/AA-600 series terminals on the network, whether registered in an IP-PBX or in P2P (Peer-to-Peer) to the IP address of the terminals. All terminals can call ADAM at the assigned number or in P2P at the IP address of the PC where ADAM is installed.



13.3. Appendix 3: Use of the AUDACITY software for audio files recording

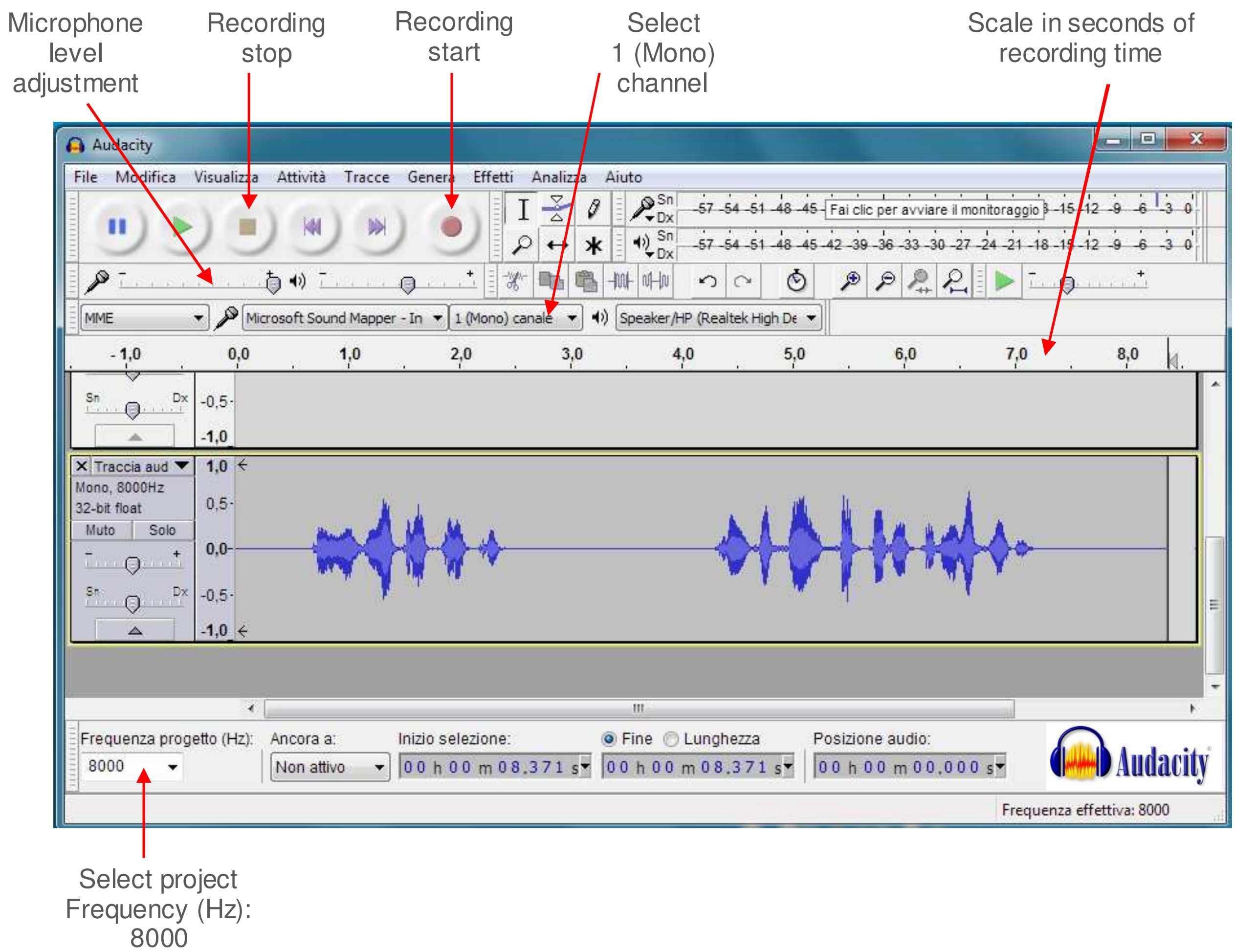
For recording your audio files, it is possible to use one of the free softwares such as AUDACITY downloadable for free from the link http://www.audacityteam.org/ remembering to record and save audio files in .WAV format at 8KHz-16Bit Mono, other audio file formats will not work with the AD600 series 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.

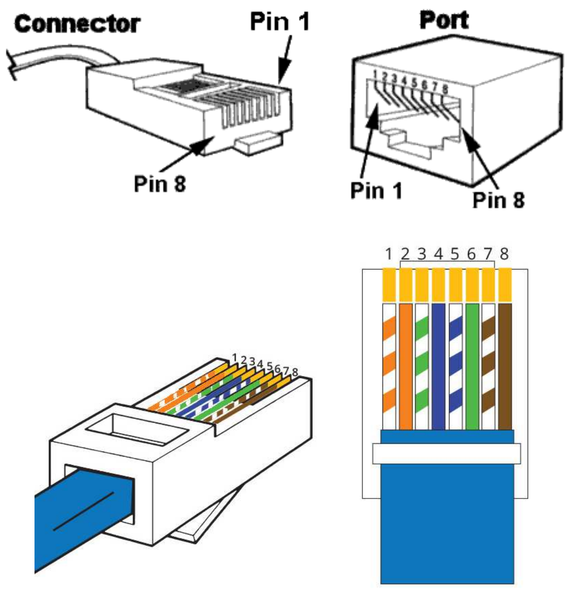
1. Select the 8,000Hz sampling rate, click Start Recording and start talking.
2. To end the recording, click on Stop recording, listen to the recorded audio, and then repeat recording if not optimal. To delete the previous recording, simply select the audio track (crawl from start to finish), press the "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.



13.4. Appendix 4: 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.

Notes