



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***

PROGRAMMING MANUAL

HW Version 1.0 – SW Version 1.1



FOR THE HARDWARE INSTALLATION OF THE DEVICES, REFER TO THE COMPLETE MANUAL

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.

Revision	Date	Revision reason	Prepared	Cecked/Approved
0	19/04/2022	First edition	MM, DP	DP, FL

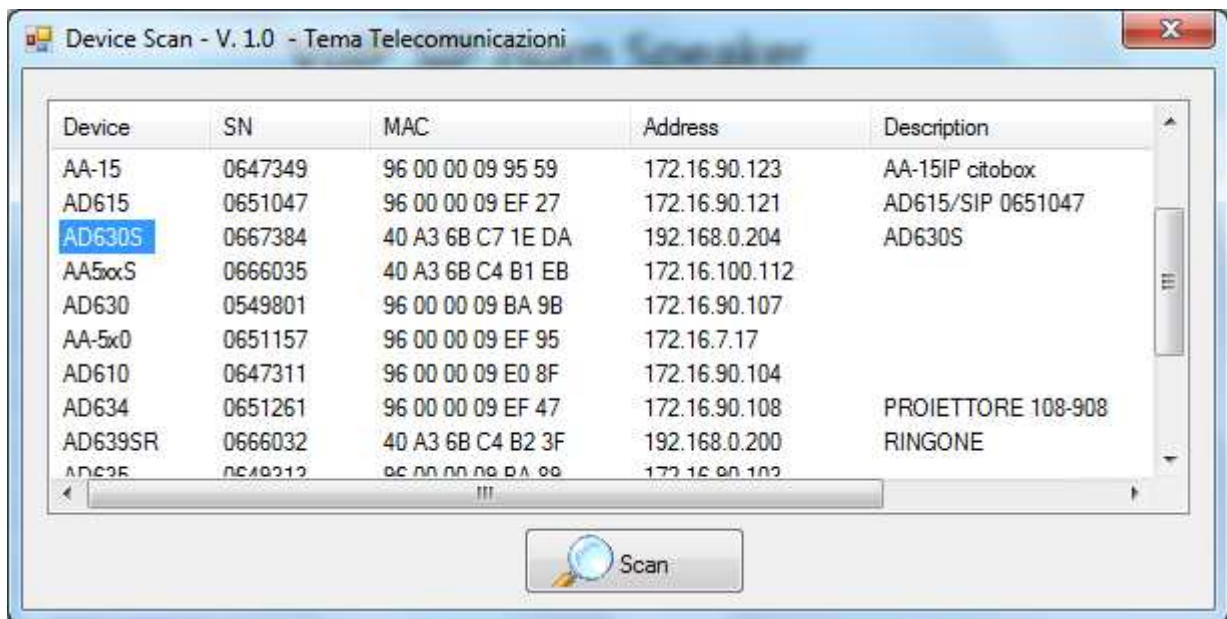
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1. PROGRAMMING

1.1. 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.

1.2. 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:

AD630S

VoIP SIP Horn Speaker

Logout

TIME: 16:50

GENERAL INFORMATION

Serial number :	0667384
MAC address :	40 A3 6B C7 1E DA
AD version :	1.1.2
RG version :	1.0.1
Mode :	Day
Stream :	239.255.25.56:8001

LAN

IP address :	192.168.0.204
Subnet mask :	255.255.255.0
Gateway :	192.168.0.1

SIP

Main account :	REGISTERED
Ringer account :	REGISTERED

← The programming manual in PDF is also available on the web server of the device.

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.

1.3. 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.

1.4. SIP Parameters

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

AD630S

VoIP SIP Horn Speaker

SERVER SIP

IP address : 192.168.0.56

Port : 5060

Domain : 192.168.0.56

Outbound proxy : 192.168.0.56

Local port : 5060

Expiration : 900

MAIN ACCOUNT

User : 553

Password : ...

ACCOUNT RINGER

User : 556

Password : ...

Ringer volume : 2

Ringer type : Ring default

Test Ring ON

Test Ring OFF

Increasing volume : ☒

Relay activation 1 : Continuous

Relay activation 2 : Burst

Replicate ring in stream : ☒


Audio stream number : 1 (select one of out streams)

Save


TIME: 23:11

- 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



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▶ Multicast

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MULTICAST AUDIO RECEIVE

Function active: ☒ Enable ☐ Disable

#1 - IP address :	239.255.15.1	Port :	8001	Volume :	0	<input checked="" type="radio"/>
#2 - IP address :	239.255.15.2	Port :	8001	Volume :	1	<input type="radio"/>
#3 - IP address :	239.255.15.3	Port :	8001	Volume :	0	<input type="radio"/>
#4 - IP address :	239.255.15.4	Port :	8001	Volume :	1	<input type="radio"/>
#5 - IP address :	239.255.15.5	Port :	8001	Volume :	1	<input type="radio"/>
#6 - IP address :	239.255.15.6	Port :	8001	Volume :	1	<input type="radio"/>
#7 - IP address :	239.255.15.7	Port :	8001	Volume :	1	<input type="radio"/>
#8 - IP address :	239.255.15.8	Port :	8001	Volume :	0	<input type="radio"/>
#9 - IP address :	239.255.15.9	Port :	8001	Volume :	1	<input type="radio"/>
#10 - IP address :	239.255.12.43	Port :	8002	Volume :	5	priority <
#11 - IP address :	239.255.25.56	Port :	8001	Volume :	7	
#12 - IP address :	239.255.30.200	Port :	8001	Volume :	7	
#13 - IP address :	239.255.30.255	Port :	8001	Volume :	7	
#14 - IP address :	239.255.35.200	Port :	8001	Volume :	7	
#15 - IP address :	239.255.35.255	Port :	8001	Volume :	7	
#16 - IP address :	239.255.40.255	Port :	8001	Volume :	7	priority >

MULTICAST AUDIO TRANSMIT


#1 - IP address :	239.255.15.1	Port :	8001
#2 - IP address :	239.255.43.2	Port :	8001
#3 - IP address :	239.255.15.7	Port :	8001
#4 - IP address :	239.255.25.66	Port :	8001
#5 - IP address :	239.255.25.1	Port :	8001
#6 - IP address :		Port :	
#7 - IP address :		Port :	
#8 - IP address :		Port :	
#9 - IP address :		Port :	

ROUTING MULTICAST (MASTER)

Function active: ☐ Enable ☒ Disable

Audio stream number : 4 (select one of above outstreams)

TIME: 16:59



→

1.5. General settings

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

- **Communication duration:** once in connection with the extension, the call is still cut down after the set time.
- **Call attempt duration:** establishes how long the internal call attempt should last, in seconds (if the called number rings but does not answer).
- **Response time:** when the AD630S is called, it answers after the set time. If it is 0, it answers immediately, if 999 it never answers.
- **Connection code:** when the device receives an incoming call, if this field is programmed, the audio is not connected until the user enters the correct code.
- **Monitor code:** 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).
- **Registration code:** 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 '***'.
- **Push to talk:** allows you to enable the microphone only after pressing a button connected to the programmed input
- **Communication volume:** sets the volume of the audio played to the external user. Possible values range from 0 (very weak) to 9 (very strong).
- **Microphone sensitivity:** 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).
- **Ring type:** select the type of ring for incoming calls
- **Ringtone volume:** 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).
- **Signal tones / Tone volume:** 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).
- **Echo Limiter threshold:** 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.
- **Announcement code:** 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.

1.6. 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:



AD630S

VoIP SIP Horn Speaker

MODE

Current mode : ☒ Automatic ☐ Day ☐ Night

AUTOMATIC mode code : *0

DAY mode code : *3

NIGHT mode code : *4

NIGHT BANDS

Day	from	to	from	to	from	to	from	to
Monday	00:00	08:00	17:30	23:59				
Thursday	00:00	08:00	17:30	23:59				
Wednesday	00:00	08:00	17:30	23:59				
Thursday	00:00	08:00	17:30	23:59				
Friday	00:00	08:00	17:30	23:59				
Saturday	00:00	23:59						
Sunday	00:00	23:59						

Save

TIME: 15:56

- 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

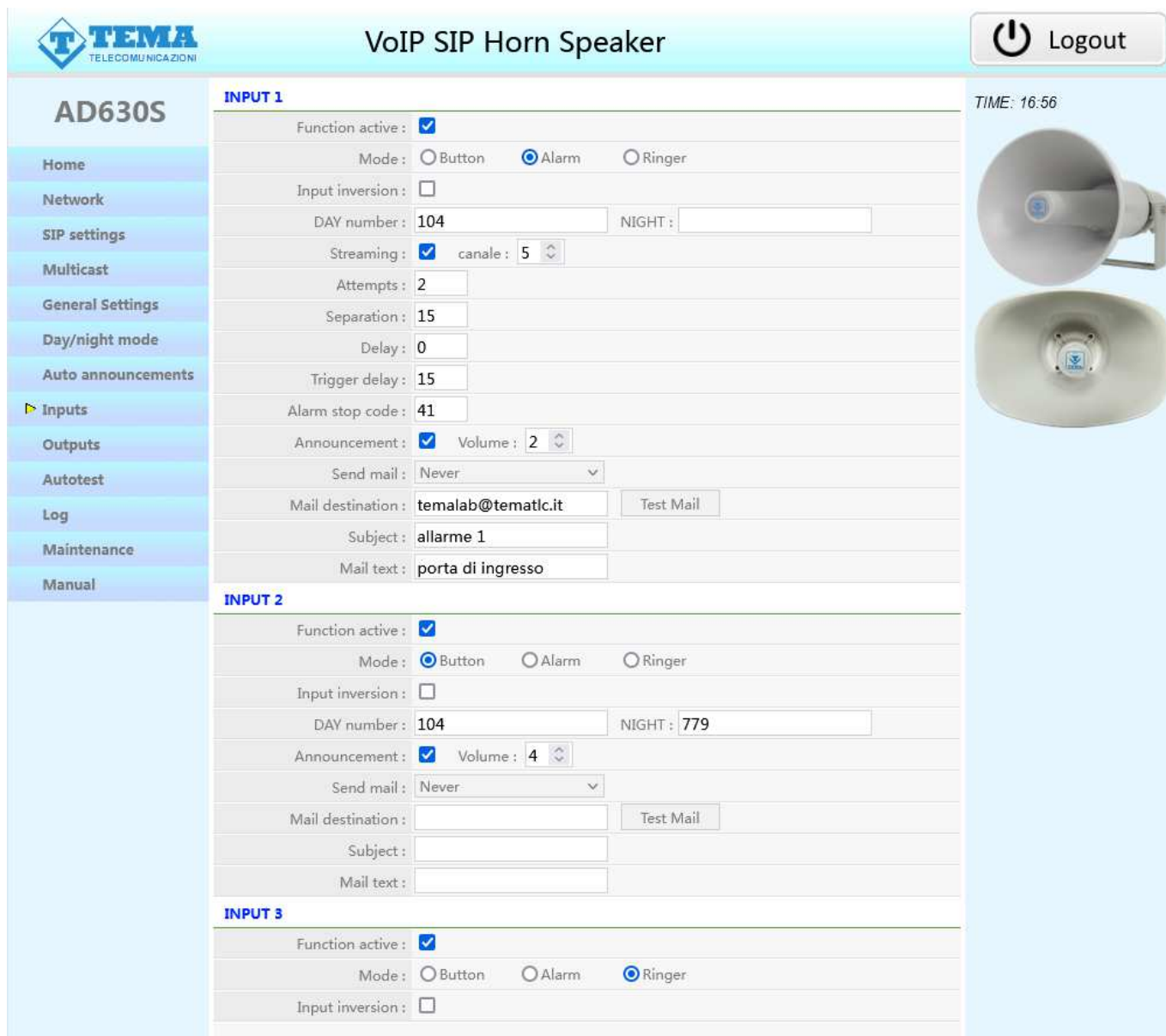
The fields in the table indicate in which time slot the system is in the night. Outside these time slots, the device is in the day. In the example from Monday to Friday the system is in the night until 8 in the morning and then from 17:30 until midnight. Saturday and Sunday it is always at night. At 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.

1.7. Inputs

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



VoIP SIP Horn Speaker

TIME: 16:56

AD630S

INPUT 1

Function active: ☒

Mode: ☐ Button ☒ Alarm ☐ Ringer

Input inversion: ☐

DAY number: 104 NIGHT:

Streaming: ☒ canale: 5

Attempts: 2

Separation: 15

Delay: 0

Trigger delay: 15

Alarm stop code: 41

Announcement: ☒ Volume: 2

Send mail: Never

Mail destination: temalab@tematlc.it Test Mail

Subject: allarme 1

Mail text: porta di ingresso

INPUT 2

Function active: ☒

Mode: ☒ Button ☐ Alarm ☐ Ringer

Input inversion: ☐

DAY number: 104 NIGHT: 779

Announcement: ☒ Volume: 4

Send mail: Never

Mail destination: Test Mail

Subject:

Mail text:

INPUT 3

Function active: ☒

Mode: ☐ Button ☐ Alarm ☒ Ringer

Input inversion: ☐

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 stream 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.

1.8. 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):

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VoIP SIP Horn Speaker

Logout

TIME: 16:56

OUTPUT 1

Mode: Electric lock

Activation code: #1

Activation time: 2

End call: ☐

OUTPUT 2

Mode: Electric lock

Activation code: #2

Activation time: 2

End call: ☐

OUTPUT 3

Mode: Electric lock

Activation code: #3

Activation time: 2

End call: ☐

Save

- **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

1.9. Maintenance

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.



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ALARM MESSAGES
TIME: 23:14

#1 :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Upload"/>	<input type="button" value="Play"/>
#2 :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Upload"/>	<input type="button" value="Play"/>
#3 :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Upload"/>	<input type="button" value="Play"/>

ANOUNCE MESSAGES

#1 :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Upload"/>	<input type="button" value="Play"/>
#2 :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Upload"/>	<input type="button" value="Play"/>
#3 :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Upload"/>	<input type="button" value="Play"/>
#4 :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Upload"/>	<input type="button" value="Play"/>
#5 :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Upload"/>	<input type="button" value="Play"/>
#6 :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Upload"/>	<input type="button" value="Play"/>
#7 :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Upload"/>	<input type="button" value="Play"/>
#8 :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Upload"/>	<input type="button" value="Play"/>
#9 :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Upload"/>	<input type="button" value="Play"/>

CONFIG

Backup and restore :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Restore"/>	<input type="button" value="Save"/>
Factory reset :				
<input type="button" value="Execute"/>				

MASTER PASSWORD

Old password :	<input type="text"/>			
New password :	<input type="text"/>			
Confirm password :	<input type="text"/>			
<input type="button" value="Save"/>				

FIRMWARE


Program files :	<input type="button" value="Scegli il file"/>	Nessun file scelto	<input type="button" value="Carica"/>	<input type="button" value="Reboot"/>
-----------------	---	--------------------	---------------------------------------	---------------------------------------




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.

1.10. 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:



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MULTICAST AUDIO RECEIVE
Function active: ☒ Enable ☐ Disable

#1 - IP address:	239.255.15.1	Port:	8001	Volume:	0	<input checked="" type="radio"/>
#2 - IP address:	239.255.15.2	Port:	8001	Volume:	1	<input type="radio"/>
#3 - IP address:	239.255.15.3	Port:	8001	Volume:	0	<input type="radio"/>
#4 - IP address:	239.255.15.4	Port:	8001	Volume:	1	<input type="radio"/>
#5 - IP address:	239.255.15.5	Port:	8001	Volume:	1	<input type="radio"/>
#6 - IP address:	239.255.15.6	Port:	8001	Volume:	1	<input type="radio"/>
#7 - IP address:	239.255.15.7	Port:	8001	Volume:	1	<input type="radio"/>
#8 - IP address:	239.255.15.8	Port:	8001	Volume:	0	<input type="radio"/>
#9 - IP address:	239.255.15.9	Port:	8001	Volume:	1	<input type="radio"/>
#10 - IP address:	239.255.12.43	Port:	8002	Volume:	5	priority <
#11 - IP address:	239.255.25.56	Port:	8001	Volume:	7	
#12 - IP address:	239.255.30.200	Port:	8001	Volume:	7	
#13 - IP address:	239.255.30.255	Port:	8001	Volume:	7	
#14 - IP address:	239.255.35.200	Port:	8001	Volume:	7	
#15 - IP address:	239.255.35.255	Port:	8001	Volume:	7	
#16 - IP address:	239.255.40.255	Port:	8001	Volume:	7	priority >

MULTICAST AUDIO TRANSMIT

#1 - IP address:	239.255.15.1	Port:	8001
#2 - IP address:	239.255.43.2	Port:	8001
#3 - IP address:	239.255.15.7	Port:	8001
#4 - IP address:	239.255.25.66	Port:	8001
#5 - IP address:	239.255.25.1	Port:	8001
#6 - IP address:		Port:	
#7 - IP address:		Port:	
#8 - IP address:		Port:	
#9 - IP address:		Port:	

ROUTING MULTICAST (MASTER)


Function active: ☐ Enable ☒ Disable

Audio stream number:

4

(select one of above outstreams)

TIME: 16:59



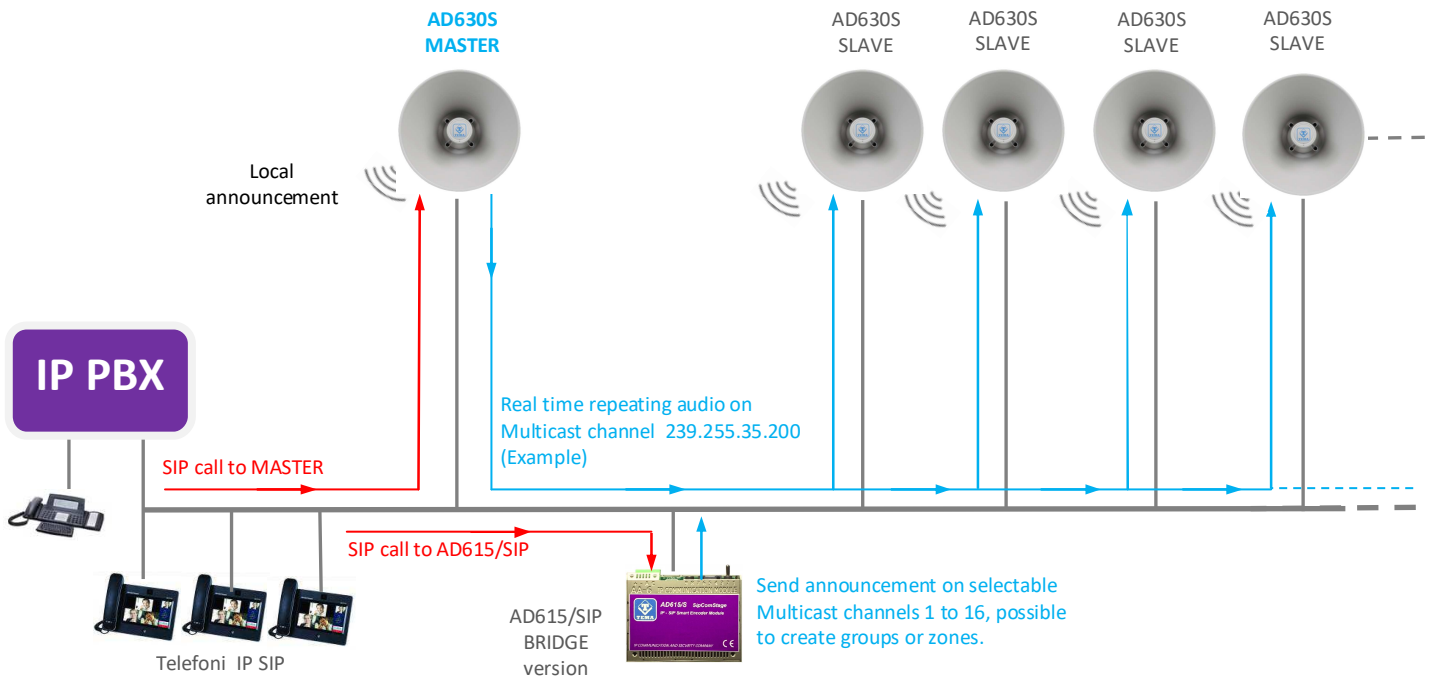
The first 9 Multicast channels are fixed and not linked to priority. You can choose only 1 among them and it allows you to choose, for example, the music you want to listen to if several audio streams are transmitted on the network at the same time.

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.

1.11. 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.

AD630S

VoIP SIP Horn Speaker

Logout

TIME: 16:59

- Home
- Network
- SIP settings
- Multicast
- General Settings
- Day/night mode
- ▶ Auto announcements
- Inputs
- Outputs
- Autotest
- Log
- Maintenance
- Manual

DAILY ANNOUNCE ACTIVATION

Start time :	16:48
Messaggio :	9
Repetitions :	1
Separation :	10

Activate now
Stop

MESSAGE VOLUME AND RELAYS ACTIVATION

Volume :	5
Relè 1 :	Burst
Relè 2 :	Burst

PERIODIC ANNOUNCE ACTIVATION

Function : ☐ Deactivated ☒ From table ☐ From file

ANNOUNCE TABLE

Giorno	ora/msg		ora/msg		ora/msg		ora/msg		ora/msg	
Lunedì										
Martedì	14:11	5	14:12	6	14:13	7	14:14	8	14:15	9
Mercoledì	18:30	1	18:31	2	18:32	3	18:33	4	18:34	8
Giovedì		1		2		3		4		1
Venerdì										
Sabato	23:36	1	23:23	2	23:24	3	23:25	4	23:26	3s3
Domenica								1		

Save

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	E	F	G	H	I	J	K	L	M	N	O
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:

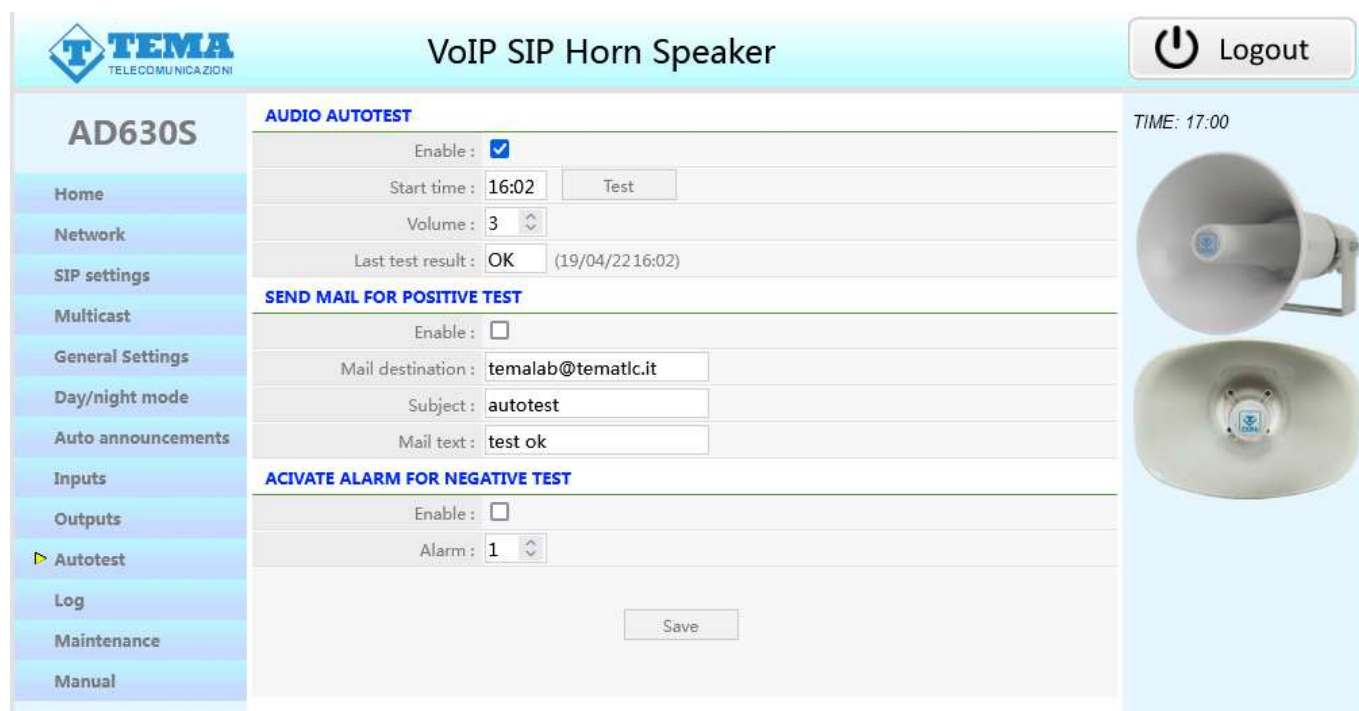
You can set the volume of the message on the local speaker and possibly the activation of relay 1 or 2.

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").

1.12. 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:



The screenshot shows the web interface for the TEMA VoIP SIP Horn Speaker. The page title is "VoIP SIP Horn Speaker". On the left is a navigation menu with options: Home, Network, SIP settings, Multicast, General Settings, Day/night mode, Auto announcements, Inputs, Outputs, Autotest (selected), Log, Maintenance, and Manual. The main content area is titled "AUDIO AUTOTEST". It contains the following settings:

- Enable:** ☒
- Start time:** 16:02 (with a "Test" button)
- Volume:** 3 (with a dropdown arrow)
- Last test result:** OK (19/04/22 16:02)
- SEND MAIL FOR POSITIVE TEST**
 - Enable:** ☐
 - Mail destination:** temalab@tematic.it
 - Subject:** autotest
 - Mail text:** test ok
- ACIVATE ALARM FOR NEGATIVE TEST**
 - Enable:** ☐
 - Alarm:** 1 (with a dropdown arrow)

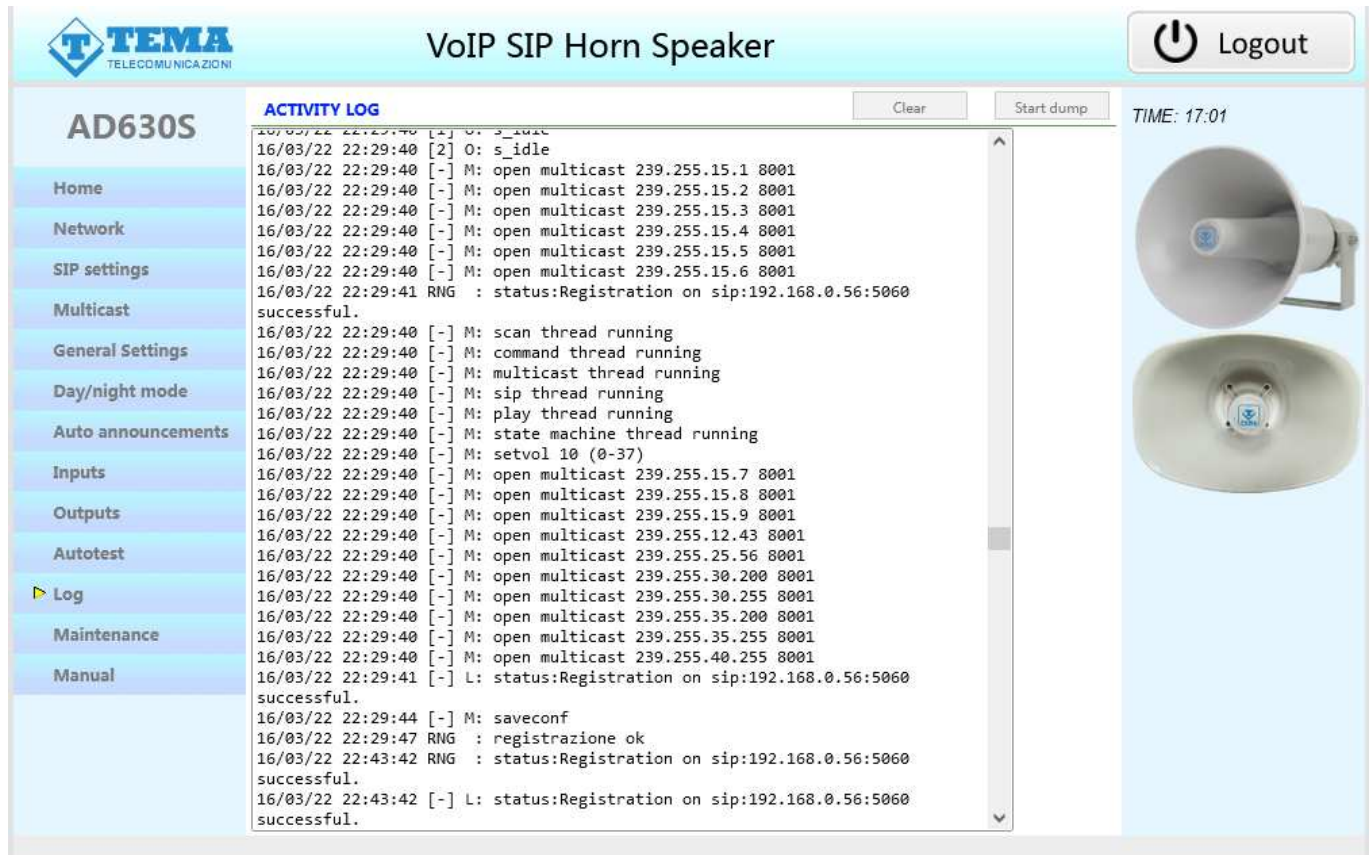
At the bottom right of the main content area is a "Save" button. On the right side of the interface, there is a "Logout" button with a power icon and a "TIME: 17:00" display. Below the time display are two images of the speaker: a side view and a front view.

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).

1.13. 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:



The screenshot displays the web interface for the TEMA VoIP SIP Horn Speaker. The interface includes a navigation menu on the left with options: Home, Network, SIP settings, Multicast, General Settings, Day/night mode, Auto announcements, Inputs, Outputs, Autotest, Log (selected), Maintenance, and Manual. The main area is titled "VoIP SIP Horn Speaker" and features a "Logout" button. Below the title, there is a section for the "ACTIVITY LOG" with "Clear" and "Start dump" buttons. The log displays a series of messages with timestamps and status indicators. On the right side, there is a "TIME: 17:01" display and two images of the speaker unit.

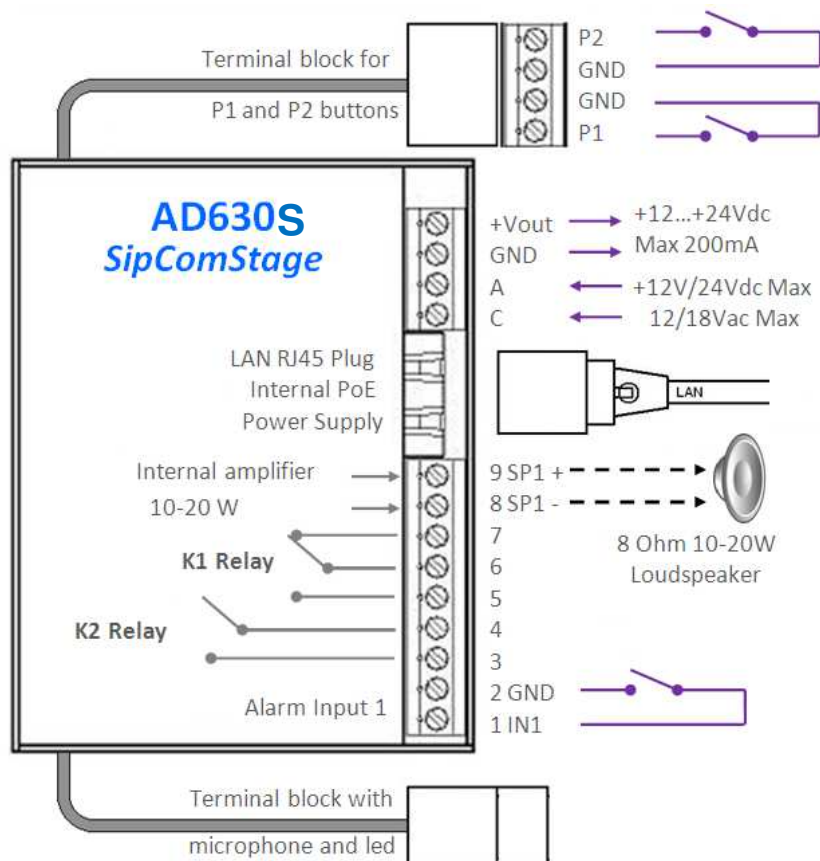
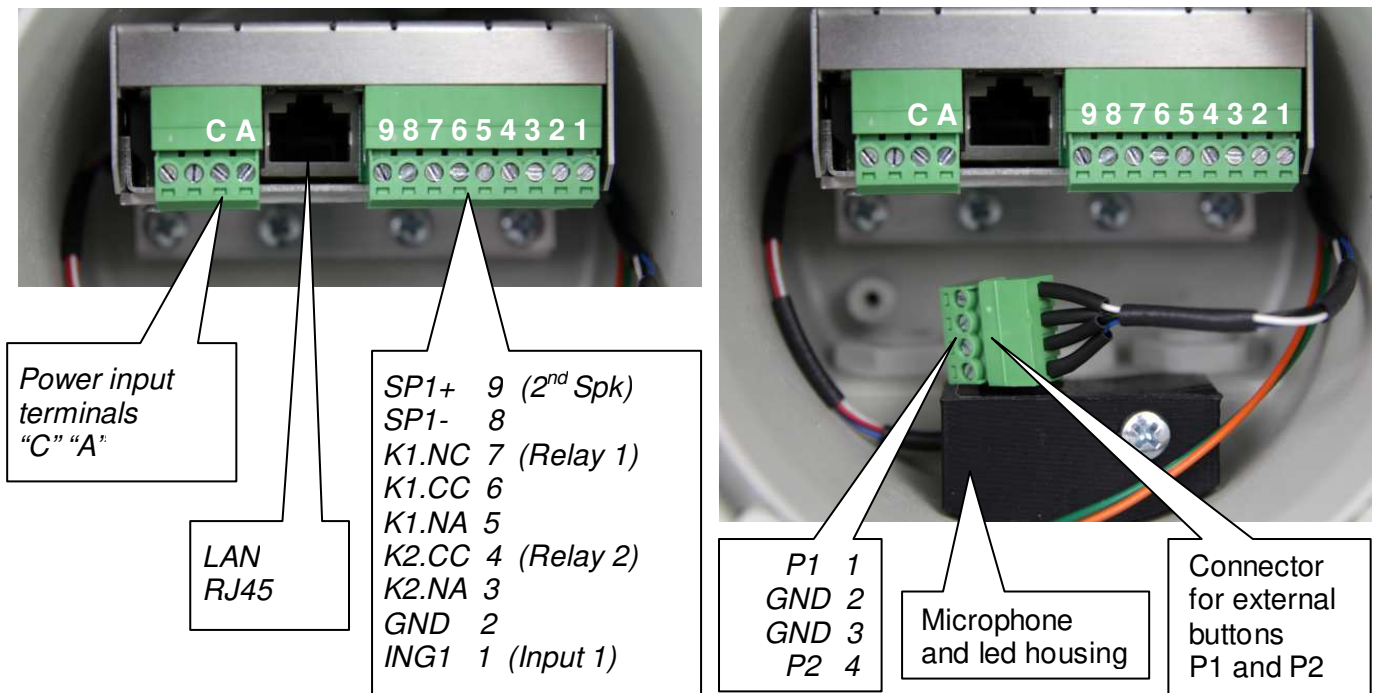
Timestamp	Status	Message
16/03/22 22:29:40	[2]	O: s_idle
16/03/22 22:29:40	[-]	M: open multicast 239.255.15.1 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.15.2 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.15.3 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.15.4 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.15.5 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.15.6 8001
16/03/22 22:29:41	RNG	: status:Registration on sip:192.168.0.56:5060 successful.
16/03/22 22:29:40	[-]	M: scan thread running
16/03/22 22:29:40	[-]	M: command thread running
16/03/22 22:29:40	[-]	M: multicast thread running
16/03/22 22:29:40	[-]	M: sip thread running
16/03/22 22:29:40	[-]	M: play thread running
16/03/22 22:29:40	[-]	M: state machine thread running
16/03/22 22:29:40	[-]	M: setvol 10 (0-37)
16/03/22 22:29:40	[-]	M: open multicast 239.255.15.7 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.15.8 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.15.9 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.12.43 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.25.56 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.30.200 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.30.255 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.35.200 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.35.255 8001
16/03/22 22:29:40	[-]	M: open multicast 239.255.40.255 8001
16/03/22 22:29:41	[-]	L: status:Registration on sip:192.168.0.56:5060 successful.
16/03/22 22:29:44	[-]	M: saveconf
16/03/22 22:29:47	RNG	: registrazione ok
16/03/22 22:43:42	RNG	: status:Registration on sip:192.168.0.56:5060 successful.
16/03/22 22:43:42	[-]	L: status:Registration on sip:192.168.0.56:5060 successful.

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.

2. HARDWARE REFERENCE

2.1. Connecting the internal module of AD630S

Inside of AD630 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)

AD630 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 alarm contact
1 ING1	Terminal block for the detection of the external alarm contact



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)

P1	Connect here the dry contact of the button named P1
GND	Reference terminal of the P1 button contact
GND	Reference terminal of the P1 button contact
P2	Connect here the dry contact of the button named P2



To terminals 1 (P1) and 2 (GND) / 4 (P2) and 3 (GND) must only be connected a relay contact or a button free from any voltage to prevent permanent damage to the device.

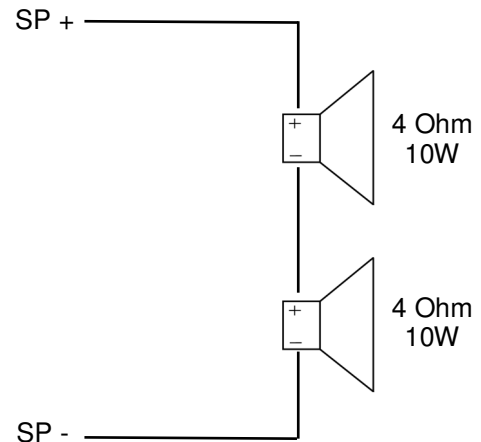
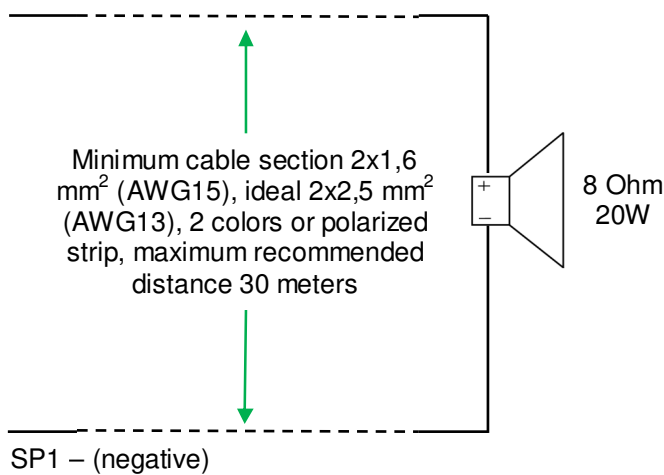
2.2. Correct connection of an additional passive speaker

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.

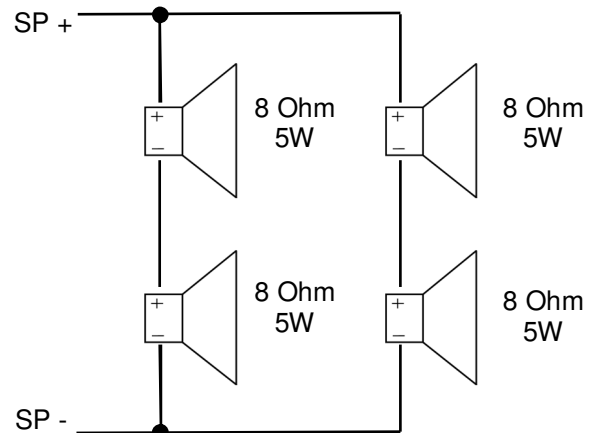
Below are some examples of mixed combinations, to serve a wider area but with distributed power.

SP1 + (positive)



Correspondence cables AWG / mm²

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 passive speakers of the same design

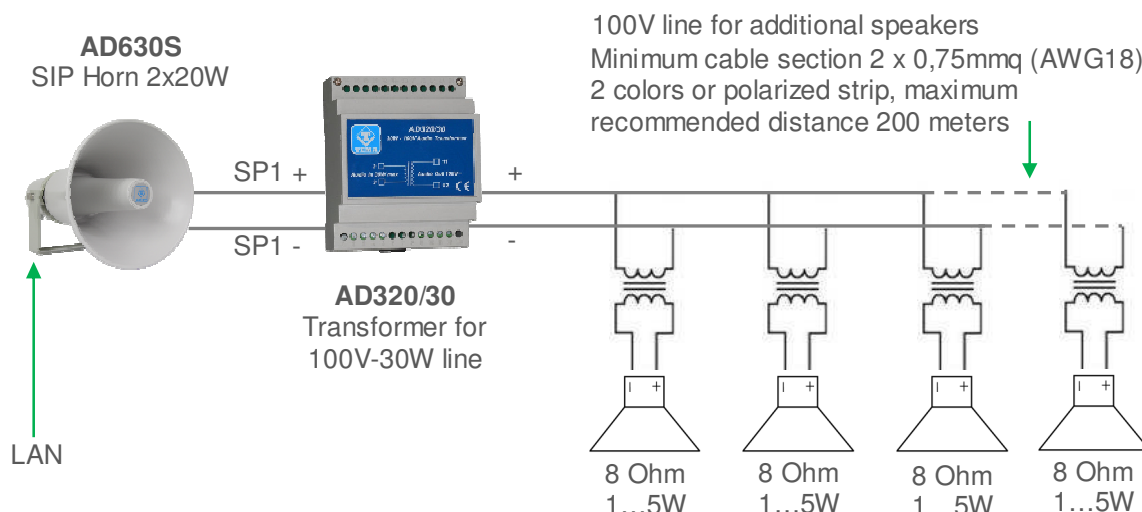
Thanks to the 2nd internal audio amplifier it is possible to connect to the active IP unit another 1 or 2 low-cost passive speakers of the same design to increase power, directionality, extend the area served, even in different rooms separated by walls, in order to reduce the cost of the plant.



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.

2.3. 100V audio line for additional speakers away from AD630S IP Horn

For special applications it is possible to generate from AD630 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 AD630. 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



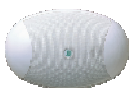
AD330/25T Horn Speaker 25W / 8Ohm with transf. 100V, sockets 25/12.5/6.25/3.12 W
response 300Hz - 7KHz, Dim. Diam. 240 x 290 mm



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)

For the complete programming of the AD630S-AD630SA devices and other details, refer to the *COMPLETE TECHNICAL MANUAL* downloadable at the links:

<https://www.tema-ipaudio.com/en/products/ip-audio/sip-speakers/ad630/>

<https://www.tematlc.it/eng/audio-ip-en.asp>

Notes