



AEQ ControlPHOENIX

**AEQ AUDIOCODECS CONFIGURATION AND CONTROL
SOFTWARE**

**TALENT, PHOENIX ALIO, STRATOS, STUDIO,
MERCURY, VENUS, VENUS V2, VENUS V3, VENUS V4 &
VENUS V4+**



**USER'S MANUAL
ED. 10/23**

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CONTENTS

1. SCOPE OF THIS MANUAL.....	4
2. APPLICATION SCOPE.....	4
3. INSTALLATION.....	5
3.1. AEQ ControlPHOENIX in a codec network: quick start guide.....	5
4. APPLICATION LAUNCH.....	9
4.1. Equipments auto-discovery: "Searching new equipment".....	9
4.2. Remote equipment discovery: "New equipment - Remote Connection".....	12
5. MAIN SCREEN.....	13
5.1. Upper menu.....	13
5.1.1. Equipment.....	13
5.1.2. Events.....	17
5.1.3. Activity.....	18
5.1.4. Connection list.....	19
5.1.5. Events List.....	19
5.1.6. Incoming calls.....	19
5.1.7. Contacts.....	20
5.1.7.1. Creating contacts.....	21
5.1.7.1.1. CALLFORWARD contact.....	26
5.1.7.2. Editing contacts.....	26
5.1.7.3. Deleting contacts.....	27
5.1.7.4. Duplicating contacts.....	27
5.1.7.5. Loading/saving contacts list.....	27
5.1.7.6. Leaving the "Contacts" menu.....	27
5.1.8. Presets.....	28
5.1.8.1. Creating presets.....	29
5.1.8.2. Editing presets.....	32
5.1.8.3. Deleting presets.....	32
5.1.8.4. Duplicating presets.....	32
5.1.8.5. Activation of presets.....	32
5.1.8.6. Leaving the "Presets" menu.....	34
5.1.8.7. Presets examples for Phoenix Venus V4 / V4+.....	34
5.1.9. Tools.....	39
5.1.10. Settings.....	44
5.1.10.1. General Application Values.....	44
5.1.10.2. Screen View.....	47
5.1.10.3. Organize.....	48
5.1.10.4. Change Installation Name.....	48
5.1.10.5. Customize Equipment Color.....	48
5.1.10.6. About.....	49
5.1.11. Close.....	50
5.2. Connection List.....	51
5.3. Event list.....	51
5.4. Lower information bar.....	51
5.5. Connection statistics.....	52
6. INDIVIDUAL CODEC CONTROL WINDOW.....	56
6.1. General configuration screen.....	58
6.1.1. Inputs configuration.....	60
6.1.2. Routing and attenuation configuration of the analog outputs.....	61
6.1.3. General audio configuration of a channel.....	62
6.1.3.1. Coding selection.....	63
6.1.3.2. Audio detectors configuration.....	64
6.1.3.3. Communications interface selection and configuration.....	64

6.1.4. Interfaces.....	65
6.1.4.1. V35 communications interface configuration (Phoenix Stratos and Phoenix Studio only).....	65
6.1.4.2. IP communications interface configuration. SIP modes.....	65
6.1.4.3. IP communications interface configuration. RTP Raw mode. Broadcast / Multicast / Multiple Unicast.....	71
6.1.4.4. ISDN (ISDN1/ISDN2) communications interface configuration (Phoenix Stratos and Phoenix Studio only).....	73
6.1.5. "Advanced" channels configuration.....	73
6.1.6. External synchronization configuration.....	75
6.2. Contacts window.....	76
6.3. Ethernet window.....	76
6.4. Miscellaneous window.....	78
6.4.1. Local panel VU-Meters display (Phoenix Stratos and Phoenix Studio only).....	78
6.4.2. Local headphones routing (Phoenix Stratos and Phoenix Studio only).....	79
6.4.3. GPI/O - Local digital in/outs functions (Phoenix Stratos, Studio, Venus, Venus V2, Venus V3, Venus V4 and Venus V4+ only).....	79
6.4.4. Ancillary - End-to-end data channel.....	81
6.4.5. Buzzer and test.....	81
6.4.6. Web user.....	82
6.4.7. Options Licensing (Phoenix ALIO and Phoenix Venus V3, V4 & V4+ only).....	82
6.4.8. User Interface permissions (Phoenix ALIO only).....	84
6.4.9. User interface locks & options (TALENT only).....	85
6.4.10. Firmware versions.....	86
6.4.10.1. Firmware Upgrade.....	87
6.4.11. Equipment date and time.....	90
6.4.12. Reboot Equipment.....	90
6.4.13. Manage configuration.....	90
6.4.14. Presets (Phoenix Venus V4, V4+ and ALIO only).....	91
6.5. Network window.....	92
6.5.1. SNMP.....	92
6.5.2. SysLog.....	92
6.5.3. Remote control.....	93
6.5.4. Protocol.....	94
6.6. Request for help from Phoenix ALIO / TALENT ("HELP" key).....	94
6.7. CALL button: establishing a call.....	95
6.7.1. Call forwarding in PROXY SIP.....	98
7. SETTING UP PHOENIX STRATOS, PHOENIX STUDIO AND PHOENIX VENUS V4+ UNITS TO USE REMOTE CONTROL.....	99
7.1. Configuration from the equipment's front panel (Phoenix Stratos, Phoenix Studio and Phoenix Venus V4+ only).....	99
7.2. Configuration from the remote control software.....	100
8. A.E.Q. GUARANTEE.....	101

1. SCOPE OF THIS MANUAL.

The information contained in this manual is valid for **AEQ** audiocodecs family: **TALENT**, **Phoenix ALIO**, **Phoenix Stratos**, **Phoenix Studio**, **Phoenix Mercury**, **Phoenix Venus**, **Phoenix Venus V2**, **Phoenix Venus V3**, **Phoenix Venus V4** and **Phoenix Venus V4+**. In the case of TALENT, Phoenix ALIO, Phoenix Stratos, Phoenix Studio and Phoenix Venus V4+, the application provides an alternative to control the equipment since these particular units have physical controls in the front panel.

This document should be considered part of the corresponding **AEQ** audiocodec user's manual that describes concepts and operational procedures. Without this information the contents presented in this document may not be fully understood.

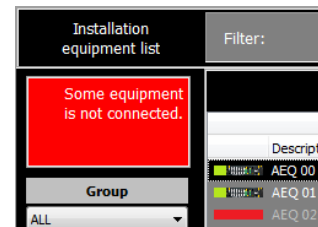
2. APPLICATION SCOPE.

The AEQ ControlPHOENIX application allows the control and management of **TALENT**, **Phoenix ALIO**, **Phoenix Stratos**, **Phoenix Studio**, **Phoenix Mercury**, **Phoenix Venus**, **Phoenix Venus V2**, **Phoenix Venus V3**, **Phoenix Venus V4** and **Phoenix Venus V4+** audiocodecs when these are connected through IP, being part of the same local area network or connected to remote networks. Local equipment are automatically discovered when the application is launched. In order to control the remote units, first of all, they need to be reachable and then the user needs to identify them by their IP address.

In order to control a **Phoenix Stratos** or **Phoenix Studio** audiocodec operating in ISDN, it is necessary to additionally provide an IP connection that is reachable from the PC where the software is installed.

This application can be installed and run on several computers that at the same time will share the control of the codecs. In the case of **TALENT**, **Phoenix ALIO**, **Phoenix Stratos**, **Phoenix Studio** and **Phoenix Venus V4+**, they also share control with the equipment's front panel. The system has been dimensioned so that each unit can be concurrently controlled by up to 4 applications.

If a fifth application tries to control the same codec, a red indication will appear in the control screen indicating that the equipment remote control is not possible as will be described later on.



The **AEQ** units can be configured to continuously attempt the connection to a pre-defined computer IP.

IMPORTANT NOTE 1: AEQ ControlPHOENIX application (from version **2.1.0.4** on) allows you to control up to **2 audiocodecs** by default. In order to add and control more than 2 units at the same time you must purchase a user license. It's possible to activate it from the window you access to by pressing the **"License"** button that appears in **"About"** option of **"Settings"** menu (see section 5.1.10.6 of this manual): in that case the license is associated with the computer where the application is installed. In order to activate that licence as well as for controlling several audiocodecs from the application, it must be done from a PC user with **administrator** rights. It's also possible to activate the license by means of a **Hardlock key**.

IMPORTANT NOTE 2: Version **2.2.1.1** (or higher) of **AEQ ControlPHOENIX** application allows you to control Phoenix audiocodecs with CPU version lower than **5.50** when the protocol used is **GEP1.0**. When protocol is **GEP1.1** (it's the protocol configured **by default** when application is installed) only audiocodecs with CPU version **5.50** or higher can be controlled. See **section 5.1.10.6** of this manual.

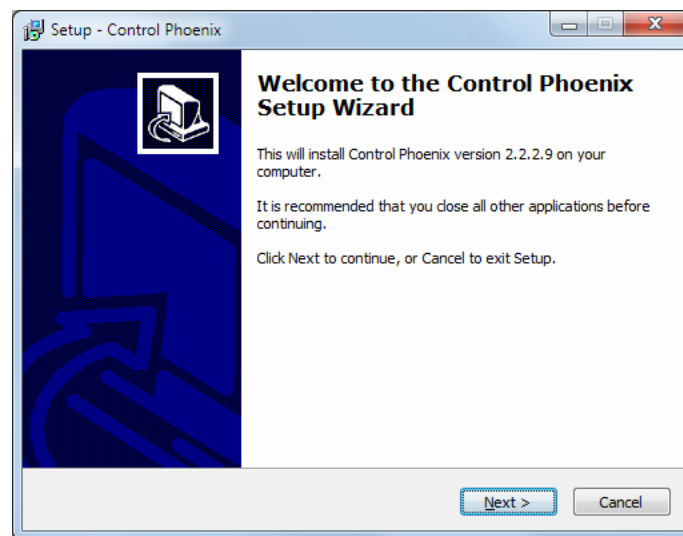
It's NOT possible to control units with version 5.50 or higher and units with previous versions SIMULTANEOUSLY. In that case, these last ones should be upgraded to version 5.50 or higher.

If you have any doubt, please consult the AEQ Technical Service (support@aeq.es).

The **AEQ ControlPHOENIX** application is certified to run on **Windows Vista**, **Windows 7** and **Windows 10** operating systems. It has also been tested on **Windows XP**, although in this case care must be taken with the Windows themes configuration as with some of them the graphical representation may not be adequate.

3. INSTALLATION.

The auto run USB key furnished with the audiocodec contains the executable file that installs the **AEQ ControlPHOENIX** application. Installing the application is simply a matter of executing this file and follow the on-screen instructions as they come up.



Once the **AEQ ControlPHOENIX** application is installed (by default, in C:\Program files\AEQ\ControlPhoenix), you can start it up by double-clicking the icon displayed on the desktop:



The application will also be accessible from the Start menu.

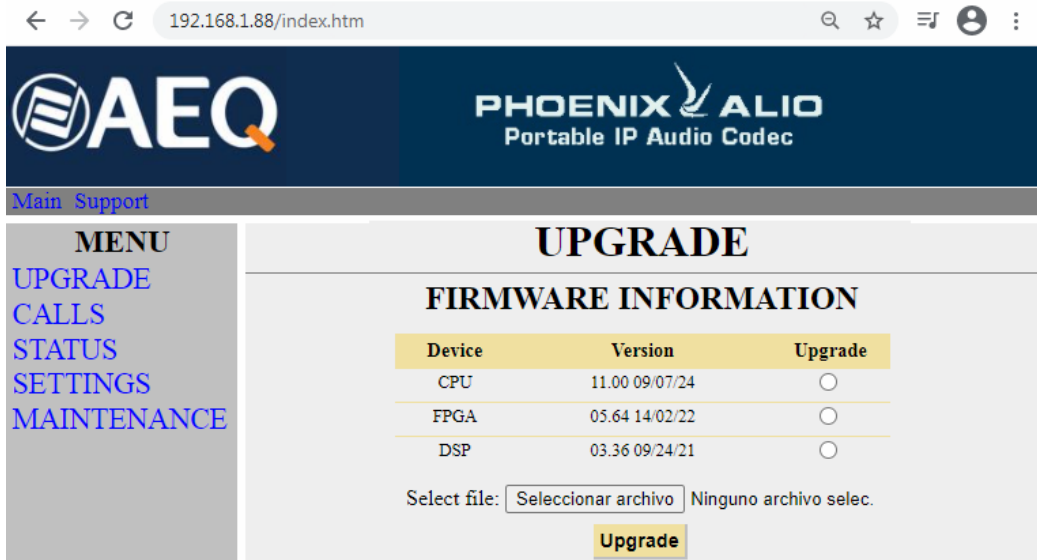
3.1. AEQ ControlPHOENIX in a codec network: quick start guide.

The procedure is as follows:

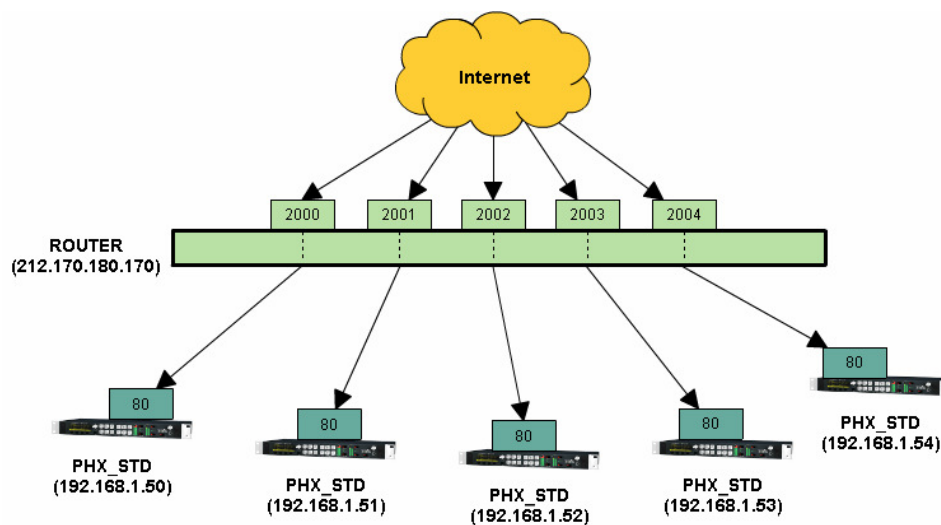
- 1) Upgrade the audiocodecs **firmware** to the last available versions. It can be accomplished through the unit's web server, that can be accessed by opening an Internet browser and typing the URL: **http://<equipment IP>/index.htm**

By default, the control IP addresses associated to the AEQ audiocodecs are the following ones:

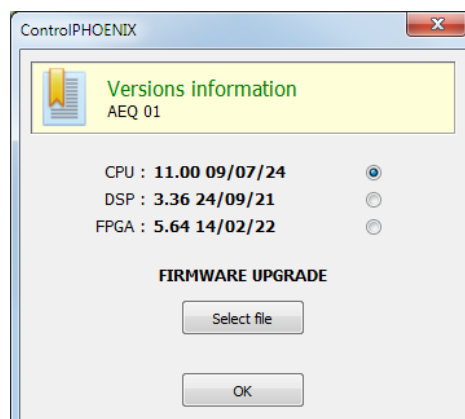
- **PHOENIX Studio:** 192.168.1.80 (port 1) and 172.26.33.80 (port 2)
- **PHOENIX Mercury:** 192.168.1.84
- **PHOENIX Venus:** 192.168.1.86
- **PHOENIX Venus V2, V3, V4 & V4+:** 192.168.1.86 (port 1) and 172.26.33.80 (port 2)
- **PHOENIX Stratos:** 192.168.1.87
- **PHOENIX ALIO:** 192.168.1.88
- **TALENT:** 192.168.1.89



If the units are accessible only through Internet, then the router in front of the equipment has to be configured: a Port Forwarding has to be assigned to port 80 of each audiocodec, assigning a different port for each of them in the router. After this, the URL to be typed in the browser to access each audiocodec has to be modified this way: **http://<router public IP>:<port>/index.htm**



The firmware upgrade can also be accomplished from the **AEQ ControlPHOENIX** application (see section 6.4.10.1 of this manual).



In the case of **Phoenix Venus V3, V4 and V4+** there is also an **AoIP** module which firmware must be upgraded by means of "**DANTE Firmware Update Manager**" application (for more information, please consult AEQ AoIP user's manual).

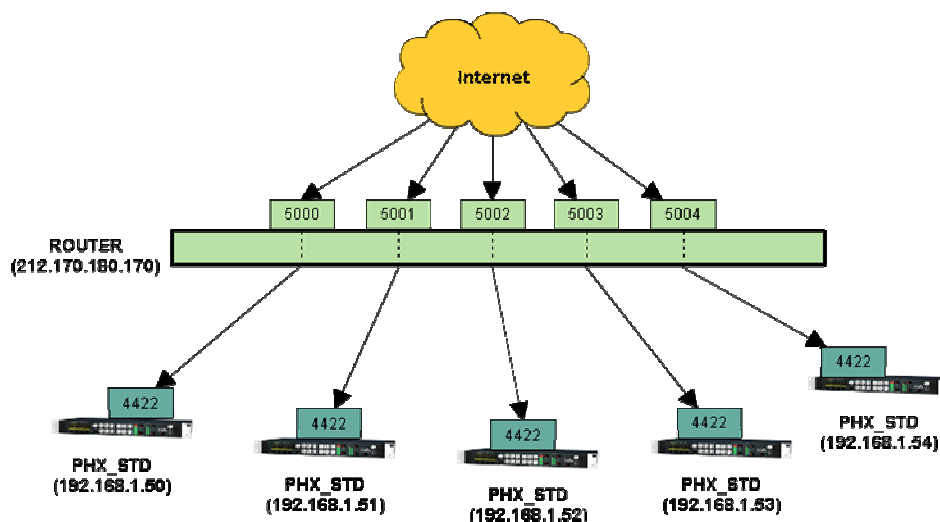
- 2) In the case of **Phoenix Stratos, Phoenix Studio and Phoenix Venus V4+**, set the audiocoders up to accept incoming remote control connections. Make sure that the "INCOMING INTERFACE" option ("IN I/F" in the case of V4+) is assigned to the Ethernet port actually in use ("ETH1" or "NET1" by default), or to "BOTH" (both interfaces, only for Phoenix Studio and Phoenix Venus V4+).

```

REMOTE CONTROL SETTINGS (R)
INCOMING INTERFACE: ETH1(Main)  INCOMING
OUTGOING INTERFACE: ETH1(Main)  1 CONN.
OUTGOING MODE: MANUAL          OUTGOING
SERVER HOST: 172.26.5.6        NOT CONN.
CONNECT
  
```

REMOTE CTRL	
IN I/F:	NET1
OUT MODE:	AUTO
OUT I/F:	NET1
SERVER:	192.168.5.59
PORT:	4422

- 3) If the connection is accomplished through Internet, the router in front of the equipment has to be configured. A Port Forwarding has to be assigned to port 4422 of each audiocoder. A different port will be assigned in the router to each unit.



- 4) Install the **AEQ ControlIPHOENIX** application on a PC.
- 5) Run the **AEQ ControlIPHOENIX** application in the PC by double-clicking the icon displayed on the desktop:



- 6) Autodiscover the equipments connected to the same local network as the PC. The application, once started, will automatically search for equipments. The user can choose whether accept the control of each audiocoder found or not.

IMPORTANT NOTE 1: AEQ ControlIPHOENIX application (from version **2.1.0.4** on) allows you to control up to **2 audiocoders** by default. In order to add and control more than 2 units at the same time you must purchase a user license. It's possible to activate it from the window you access to by pressing the "**License**" button that appears in "**About**" option of "**Settings**" menu (see section 5.1.10.6 of this manual): in that case the license is associated

with the computer where the application is installed. In order to activate that licence as well as for controlling several audiocodecs from the application, it must be done from a PC user with **administrator** rights. It's also possible to activate the license by means of a **Hardlock key**.

IMPORTANT NOTE 2: Version **2.2.1.1** (or higher) of **AEQ ControlPHOENIX** application allows you to control Phoenix audiocodecs with CPU version lower than **5.50** when the protocol used is **GEP1.0**. When protocol is **GEP1.1** (it's the protocol configured **by default** when application is installed) only audiocodecs with CPU version **5.50** or higher can be controlled. See **section 5.1.10.6** of this manual.

- 7) Adding equipment in other networks. This is a different method to manually add units to be controlled by the PC: simply indicate its IP address and control port (4422 by default). If the equipment is in another network and Port Forwarding has already been accomplished in its router or switch (see step 3), then the indicated port may not correspond to 4422. The port may have been specifically assigned by the network administrator for each unit. In order to be able to connect this way, the audiocodec public IP address must be specified, and it should be static.

In situations where it is necessary to control an equipment that is in another network (through the Internet, for example) and also the public IP of that equipment is not fixed or recognized (for example, when they are located behind a home router with Dynamic IP or connected through a 3G/4G router), it is convenient to use the "incoming" connection mode, in which the codec itself connects to the control PC, which must have a fixed public IP. A Port Forwarding on the router after which that PC is from TCP port 4422 to the private IP of the controlling PC should be carried out. In this case, the equipment must be preconfigured in advance - for which it is necessary to connect in the normal way with a PC located on the same local network - following the instructions in section 4.2 of this manual.

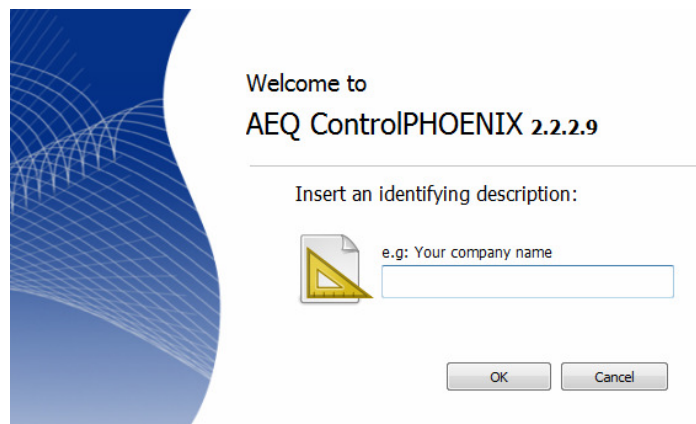
- 8) Once the PC application has been able to establish a connection with the equipment, it is possible to start controlling the registered equipments. A window including a screen capture of the units will be displayed.

4. APPLICATION LAUNCH.

When double-clicking the icon displayed on the desktop:



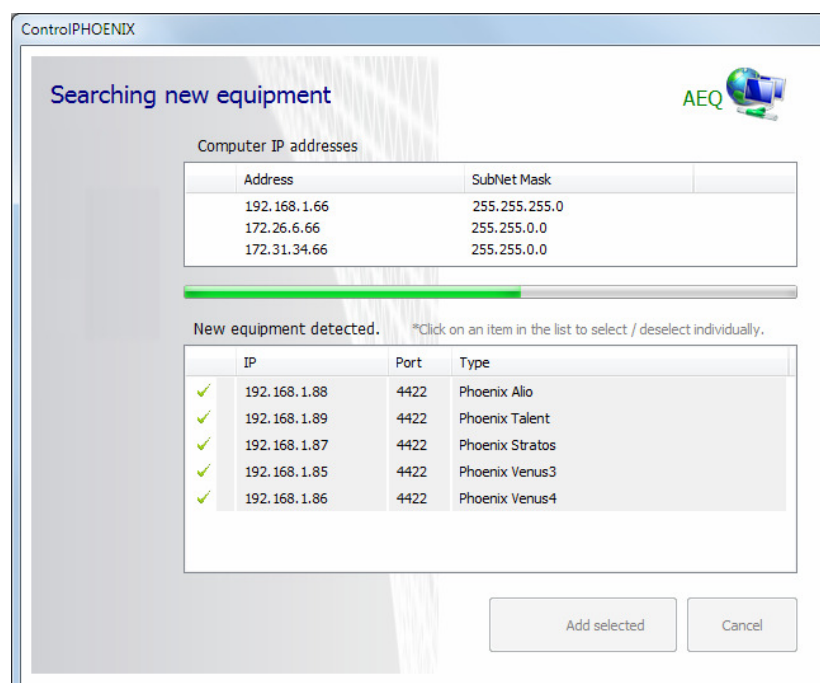
The following welcome screen will appear:



It includes the **"Insert an identifying description"** field for an identifier name. We recommend that this field is filled in with the company name. The automatically detected equipment will be identified by this name followed by consecutive digits.

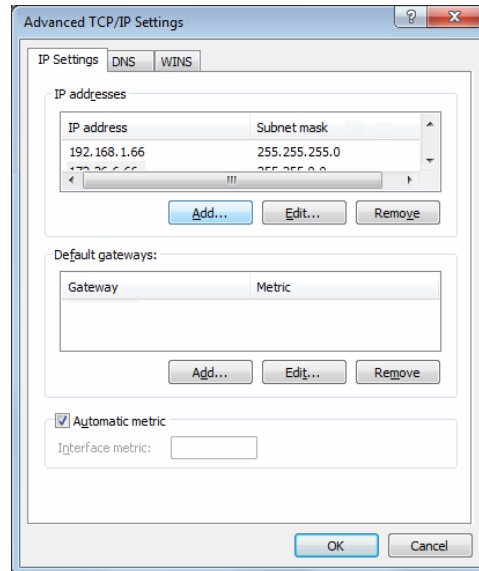
4.1. Equipments auto-discovery: "Searching new equipment".

Next, an equipments auto-discovery window will appear:



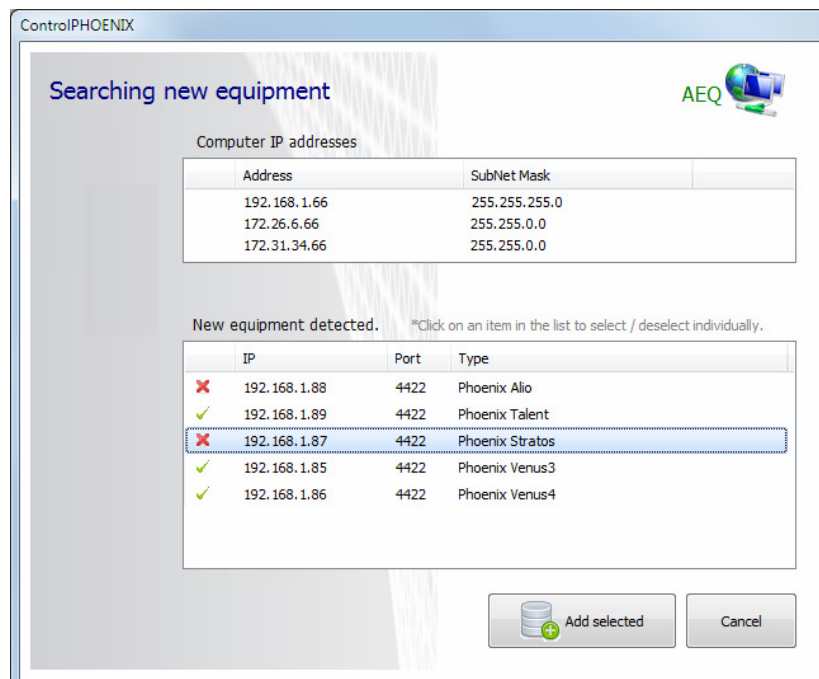
This window presents, at the top, the previously established name (AEQ in the example). Next, the computer's IP address/es and network mask/s are displayed.

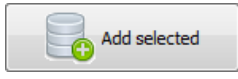
IMPORTANT NOTE: In order for the equipments to be discoverable, all audiocodecs must have an IP address within the **range** defined by the computer's IP address and network. If this is not the case, these parameters must be adjusted in the PC operating system (advanced TCP/IP configuration) or alternatively, on the actual equipment (only for **Phoenix ALIO**, **Phoenix Stratos**, **Phoenix Studio** or **Phoenix Venus V4+**) through the equipment control interface in the front panel.

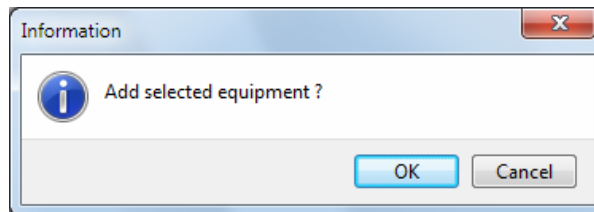


If all this has been correctly accomplished, the units will be accessible to the application and they will appear in the **"New equipment detected"** list at the bottom part of the autodiscovery window screen. By default, the name that will be assigned to the units will be the previously defined identifying name followed by consecutive digits (AEQ 00, AEQ 01, AEQ 02... in the example).

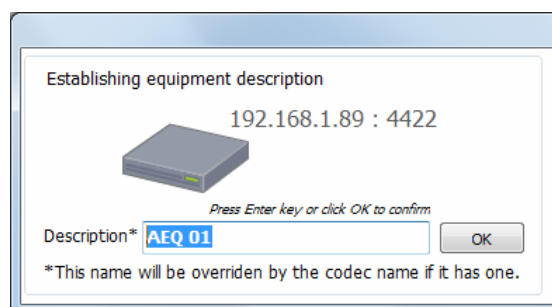
When we click on that list, we can select the equipments that we want to control and ignore the **AEQ Phoenix** audiocodecs that we do not require to be controlled by this computer or its operator. By clicking with the left mouse button on any audiocodec from the list you can select it ✓ or ignore it ✗.



By clicking on  button we will be prompted for selection confirmation:

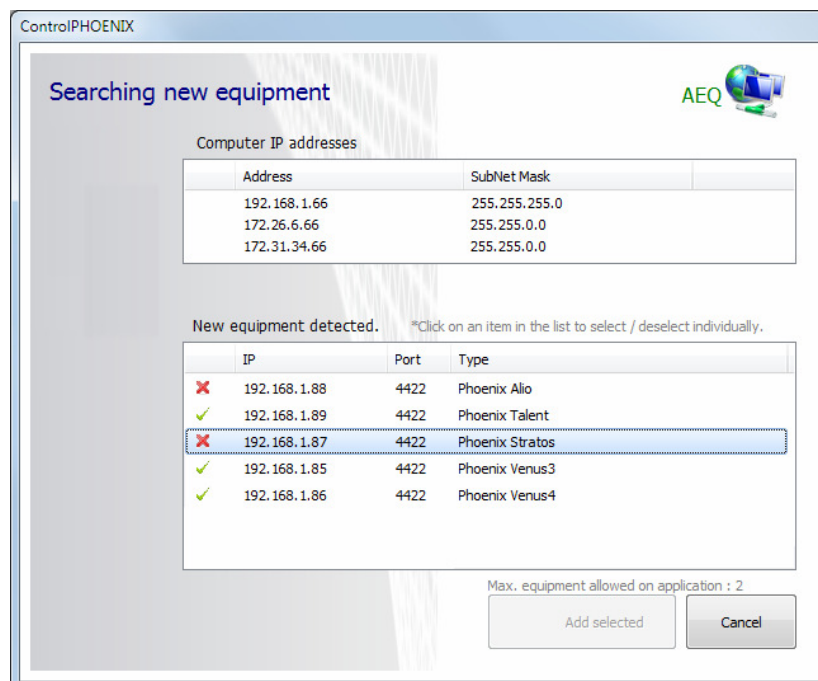


After confirmation, several consecutive screens will appear (as many as audiocodes have been selected) where you can confirm the name assigned to the unit by default or define a new one (as indicated in the window, that name will be ignored and replaced by the codec's internal name if it already has one assigned):



Whenever we launch the application and it detects that there are audiocodes reachable that weren't accepted for control before, the autodiscovery window will appear again so we can decide whether we want to add them to the pool of codecs to control by this computer.

IMPORTANT NOTE: AEQ ControlPHOENIX application (from version **2.1.0.4** on) allows you to control up to **2 audiocodes** by default. In order to add and control more than 2 units at the same time you must purchase a user license (see section 5.1.10.6 of this manual). If there's no license, the "Add selected" button will appear deactivated until a number of audiocodes equal to or less than 2 is selected (as long as there are no units already defined in the application):

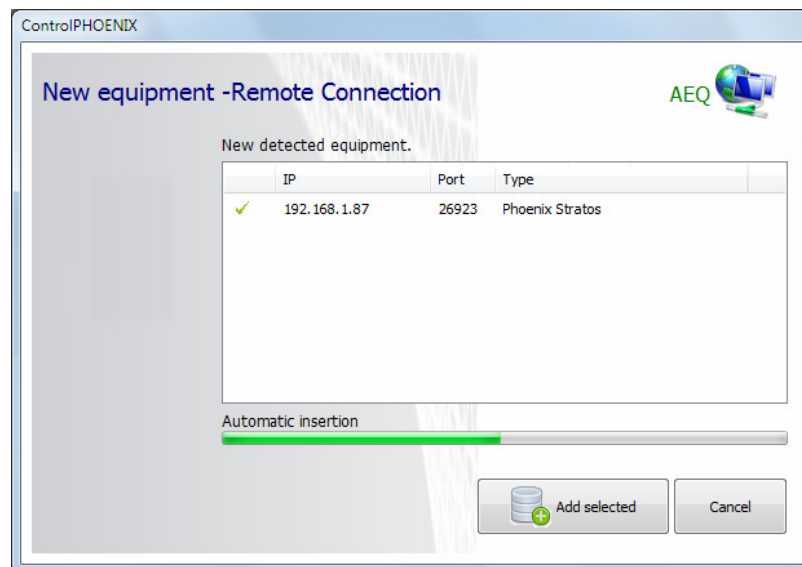


4.2. Remote equipment discovery: "New equipment - Remote Connection".

In chapter 4.1 the process of automatic discovery of equipment in compatible sub networks was described. There is an alternative procedure where the application doesn't look for equipment, but it is the equipment itself that looks for applications within its compatible sub-networks addressing the need to be controlled. This means that the equipment effectively connects to the application remotely.

This is a very useful method when the codec is connected to the Internet where its public address is not static (for example, in some home connections or mobile ones through 3G/4G routers). In these cases, it is easier to establish the control connection towards the PC, that may be located in a studio which public IP address is static (It is required to open port 4422 towards the local IP of the control PC in that studio's router).

In this case, the software will show the following window:



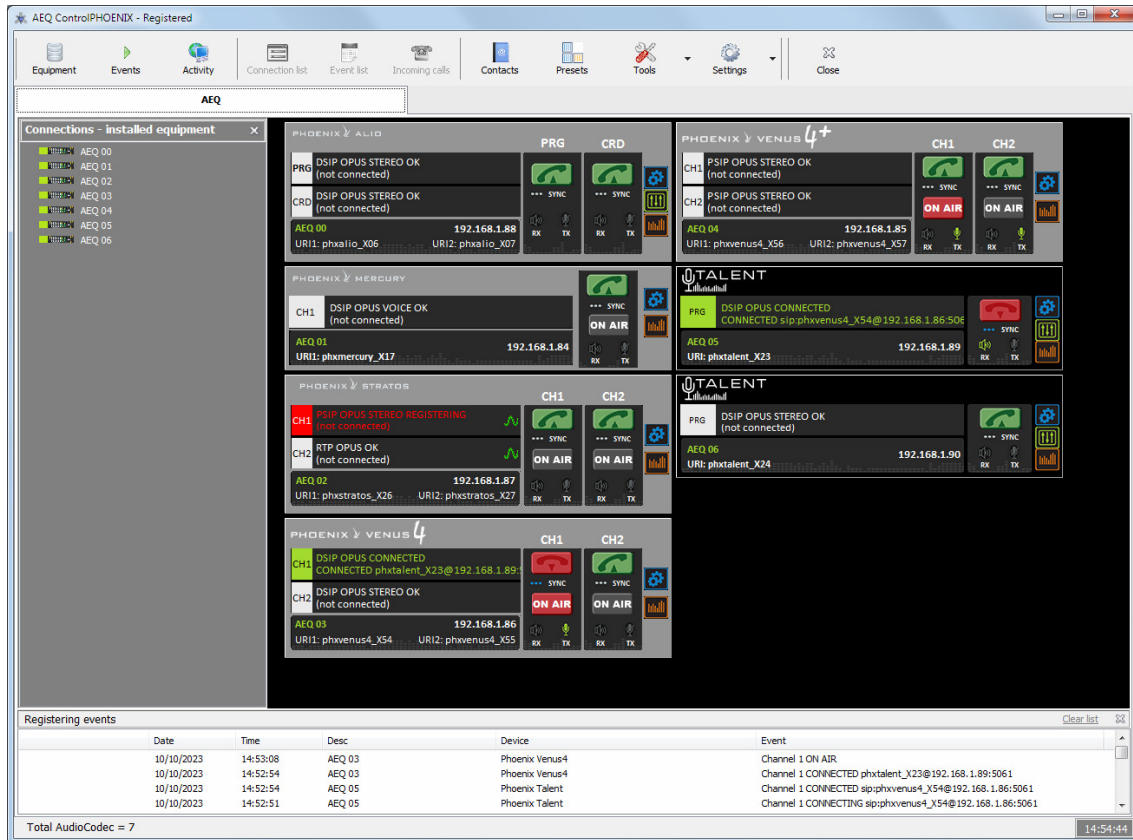
This window warns the operator that a unit wants to connect to the system. If the "**Add selected**" button is clicked, it will be incorporated immediately. If "**Cancel**" button is clicked, the connection will be denied. Please note that if no action is taken, after the time represented by the "**Automatic insertion**" progress bar expires, the equipment will be automatically added for remote control. This allows for unattended incorporation of units to the control system.

The equipment added by this automatic connection procedure will be marked by a green label "**BY REMOTE CONNECTION**" so in the future it can be removed from the control list if we no longer want to control it from this application (see section 6 of this manual). It's possible to configure whether remote connection of units is allowed or not (see section 5.1.10.1 of this manual).

IMPORTANT NOTE: AEQ ControlPHOENIX application (from version **2.1.0.4** on) allows you to control up to **2 audiocodecs** by default. In order to add and control more than 2 units at the same time you must purchase a user license (see section 5.1.10.6 of this manual).

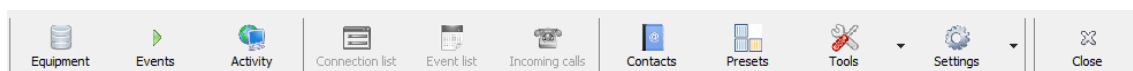
5. MAIN SCREEN.

Next, the main application screen is shown, including the detected codecs. Later on, the dimension of the symbols can be adapted to the number of codecs that have been discovered for a more convenient view:

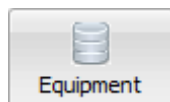


5.1. Upper menu.

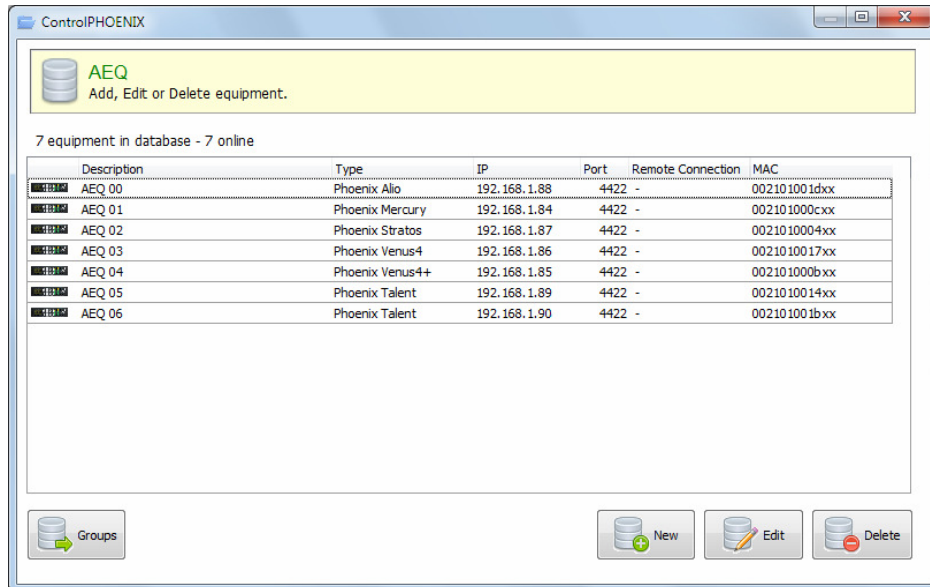
At the top of the main screen, a menu appears showing the following options:



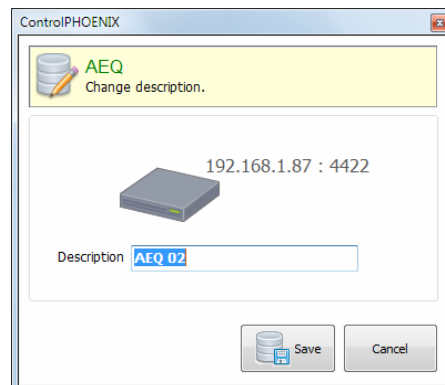
5.1.1. Equipment.



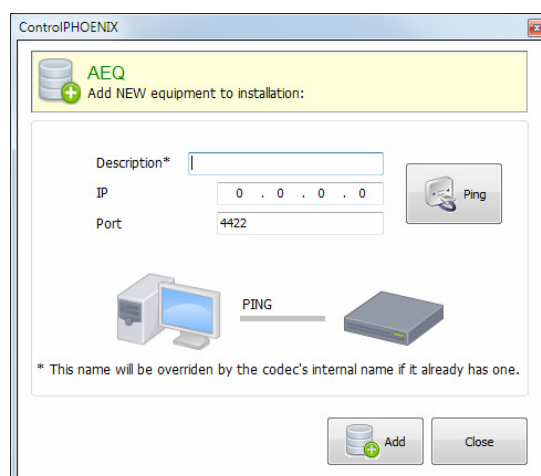
When you press this button, a window that shows a list of the installed equipments will appear:



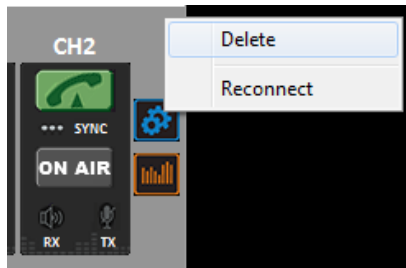
By clicking on one of the units and pressing the **"Edit"** button, you can customize its description.



By pressing the **"New"** button you can add a codec that has not been discovered, indicating a description and a route (IP address and port). You can also test this socket by issuing a **"Ping"** command by its corresponding button. As indicated in the window, that description will be ignored and replaced by the codec's internal name if it already has one assigned.



Additionally, you can remove a codec from the control list when you don't want to control it anymore by simply selecting it and clicking on **"Delete"** button. Confirmation is requested.

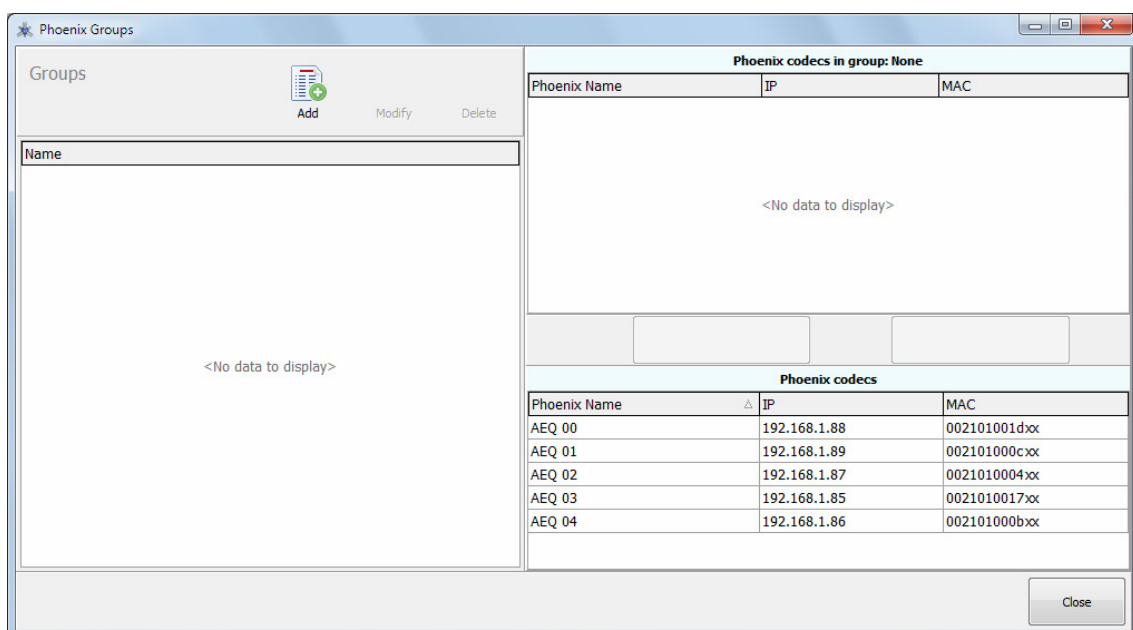


You can also remove any codec from the list by right-clicking on it in the main screen and selecting the **"Delete"** option that appears. Confirmation is also requested.

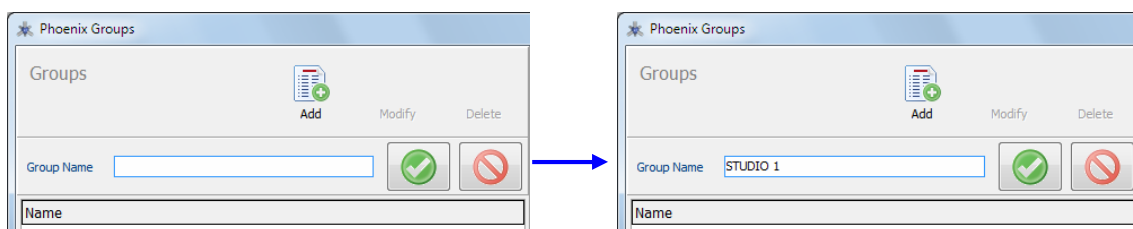
From here you can also force a reconnection with the unit by selecting the **"Reconnect"** option.

By pressing the **"Groups"** button you can create groups of units in order to select later the group you want to visualize in "List items" view mode (list of units: see section 5.1.10.2 of this manual)

When you press this button, a window will appear that allows you to "Add", "Modify" or "Delete" groups:

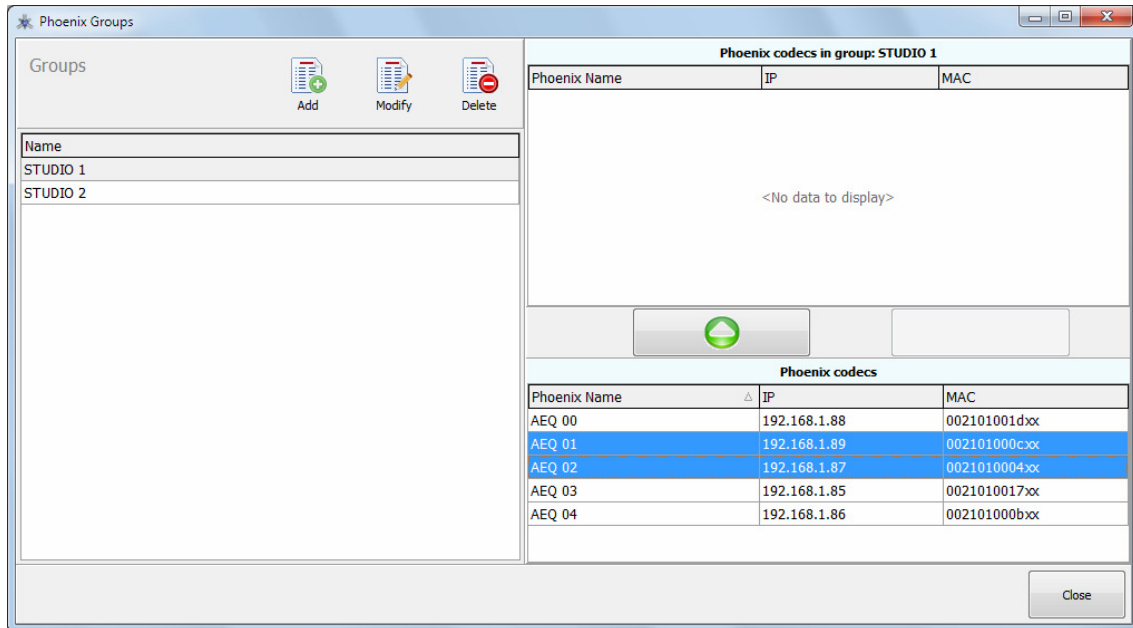


When you press the **"Add"** button, a field will appear that allows you to introduce the name of the new group:

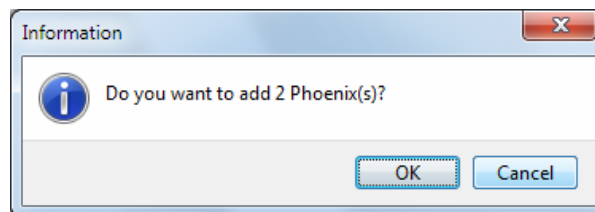


Once the desired group (or groups) is created, you should select it in the list on the left part of the screen and then choose the unit or units that will be part of that group. In order to do that, you must select them in the "Detected Phoenix codecs" list, at the right bottom part of the

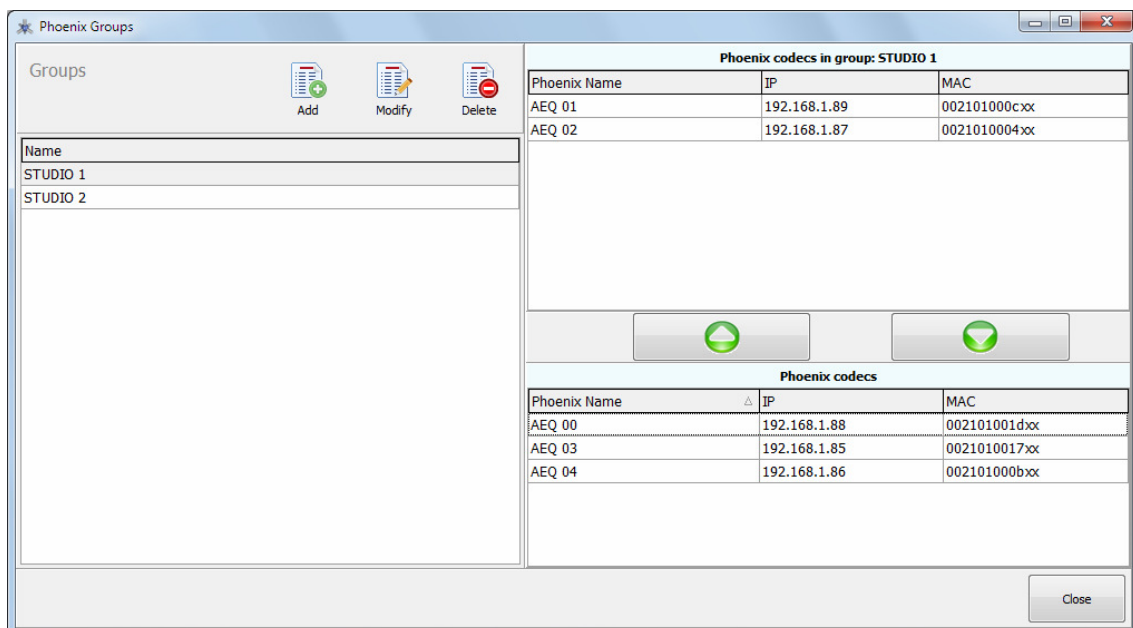
screen, and add them to the group by means of the  button.



Then a confirmation window will appear:



After confirming, the selected units will appear in the list at the right top part of the screen ("Phoenix codecs in group: STUDIO 1" in the example):



You can remove one or several units from a group by means of the

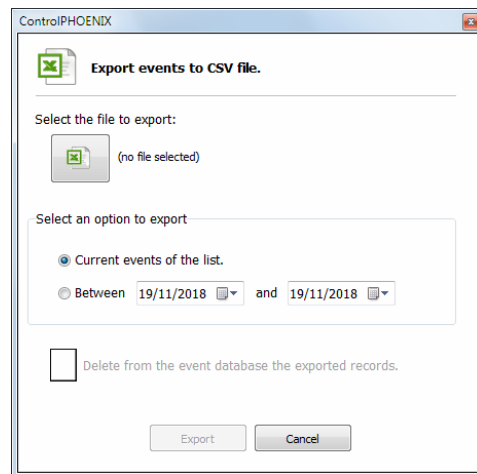


button



The  button allows you to export to CSV format the events list, in order to filter and process the information more conveniently.

In this window we can select the file name, the dates range to be exported and whether we want to remove the information from the database after being exported or not.



NOTE: In setups with a large number of installed audiocodecs and/or intense activity, it is recommended that this logging mode is deactivated in order to prevent a potential reduction of the performance as the database grows. In these cases is more useful to generate daily logs in .txt format.

In order to do this:

- 1) Delete the events list stored in the database as explained before.
- 2) Close the application.
- 3) Edit "ControlPHOENIX.ini" file, located in the same folder where the application is installed, and change the following lines:

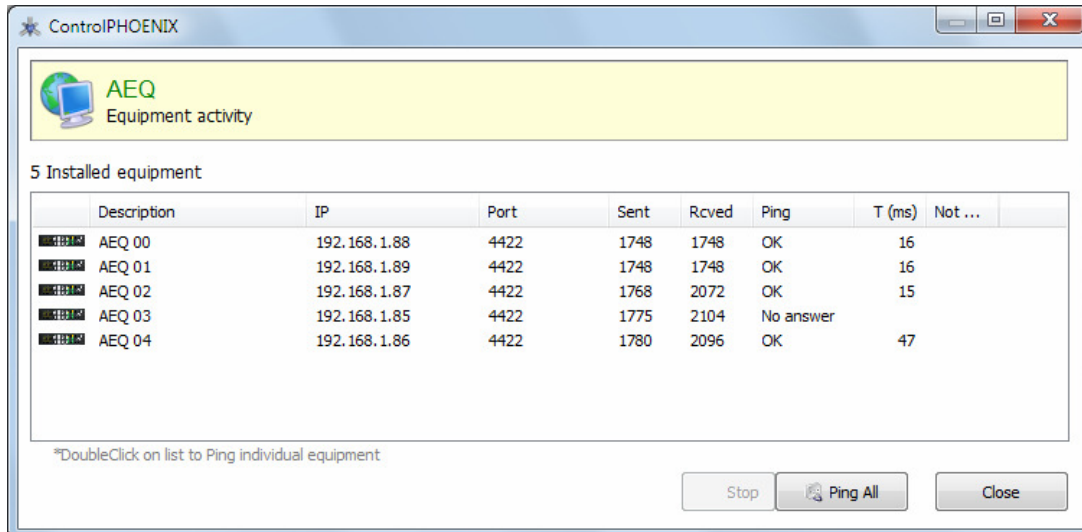
```
[LOG]
LogMode=1
TX_RX=0
TEXTEVENTS=1
DBEVENTS=0
```

From now on, once the application is reopen, a .txt file will be created daily in the "log" folder, containing detailed information about all the events produced. This is a convenient way to search long-term problems without reducing the performance of the software when many entries are logged.

5.1.3. Activity.



This button provides access to the activity control window, where you can monitor the network activity for each unit and we can also test the connection by issuing **ping** commands, either individually (by double-clicking on the unit) or to all the equipment (by means of "**Ping All**" button). The "Ping" column shows this process result and the "T (ms)" column shows the response time.

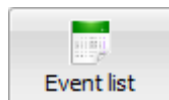


5.1.4. Connection list.



This option shows the list of connections window described in paragraph 5.2 in case it was previously hidden.

5.1.5. Events List.

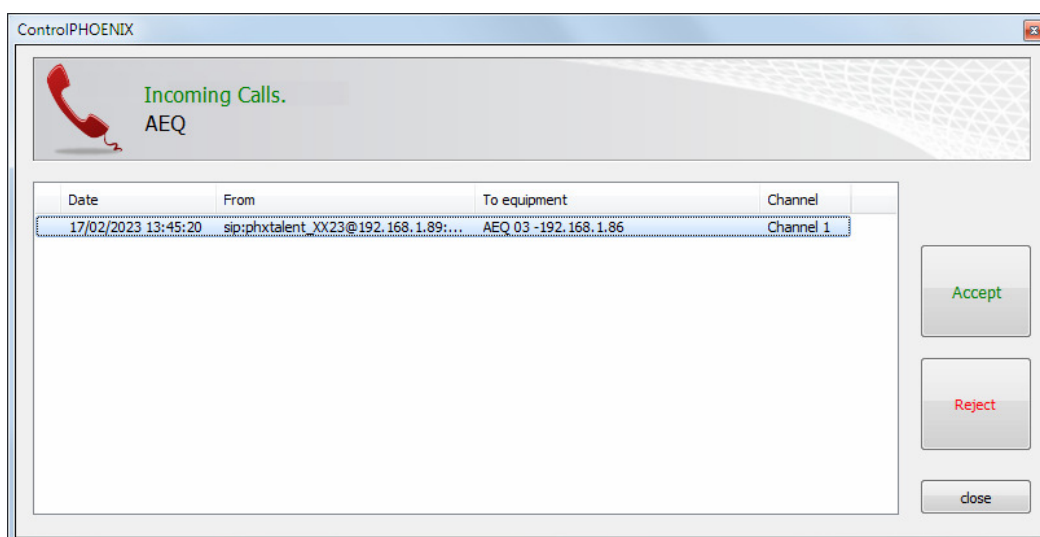


This option shows the list of events window (further described in paragraph 5.3) in case it was previously hidden.

5.1.6. Incoming calls.

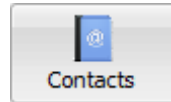


When one or more codecs of the system receive a call, an incoming calls list window appears providing the user the possibility to answer or reject each of them:

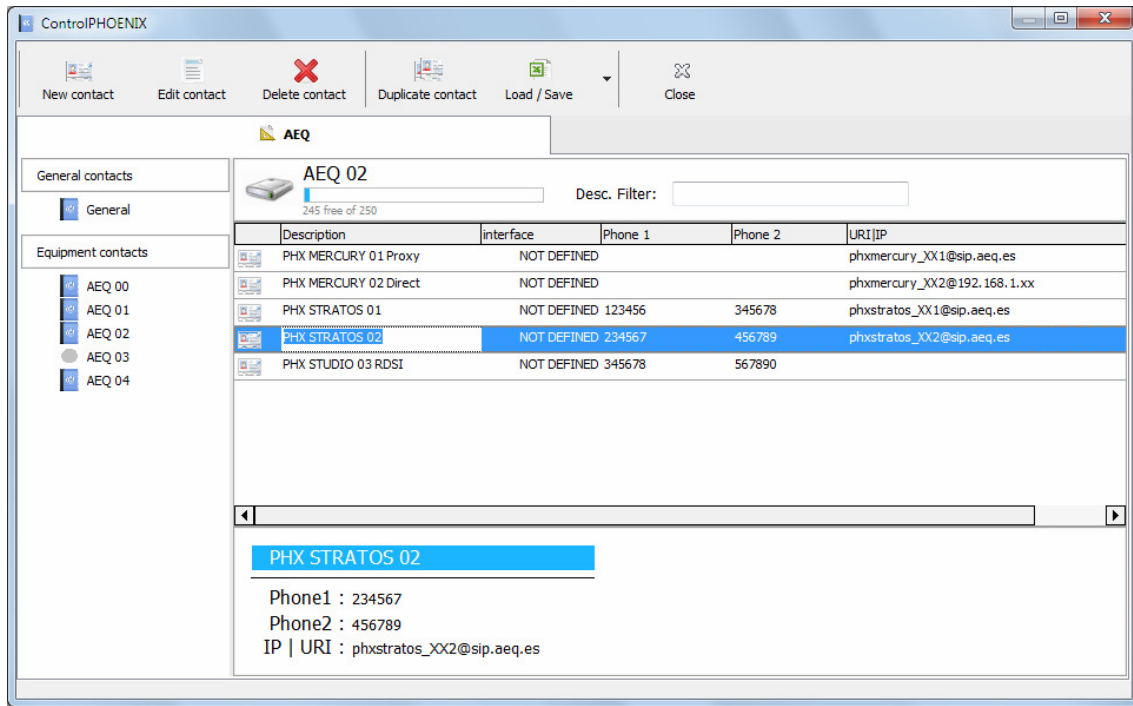


When that window is closed (by means of "close" button) and there are still unanswered calls, the "Incoming calls" icon of the upper menu changes its aspect from shaded grey to a variable color in order to signal this situation (by pressing it the incoming calls list window appears again).

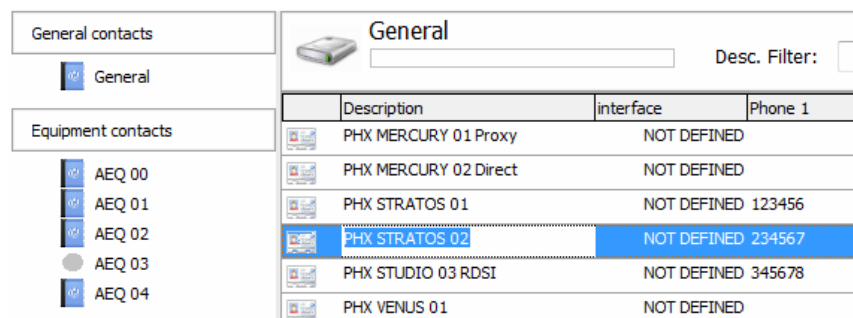
5.1.7. Contacts.



Shows the main contacts window, that manages the general and individual call books or agendas for the codec network.



- **General contacts:** the left part of the screen shows the general agenda that gives access to the contacts list that is stored in the PC where the application is installed.
- **Equipment contacts:** the left part of this area shows also the individual agendas of the equipments defined in the system, that give access to the contacts list that is stored into each equipment internal non-volatile RAM. The equipments that are connected to the system appear identified with the  symbol and those that are disconnected appear with the  symbol.
- **Contacts detail:** at the right part you can see a table with the contacts associated to the general agenda or to the codec selected from the previous list. On top you can see the "General" indication or the name of the selected codec ("AEQ 02" in the example) and, in that case, the number of available contacts (of a total of 250 for each unit). The "Desc. Filter" box allows you to filter the list by the characters that appears in "Description" field. Below that, you can see the contacts list; one line is open for each contact with the following fields: description (name), interface, telephone #1, telephone #2 and IP or URI.



When you select any of those contacts, at the lower part of the window, below a blue band, you can see the details (name, numbers, URI) of the selected contact.

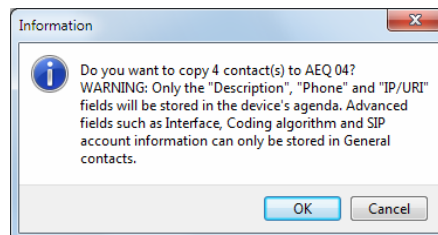
PHX STRATOS 02

Phone1 : 234567

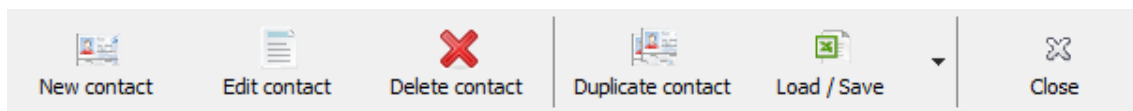
Phone2 : 456789

IP | URI : phxstratos_XX2@sip.aeq.es

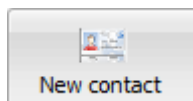
You can **copy contacts between books** through simple "drag and drop". You must select the contacts to be copied from the general agenda or a codec list (on the right part of the screen) and drag and drop it to the desired codec or the general agenda (on the left part of the screen). A confirmation window will appear:



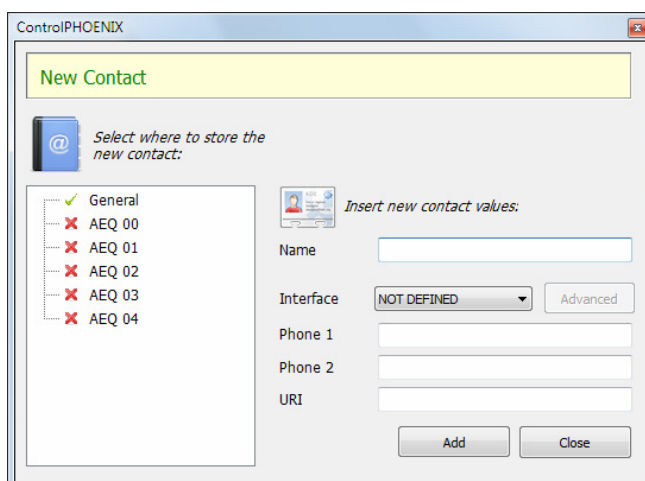
The upper menu of the main contacts window includes buttons to add, edit and delete contacts, load/save the agenda from/to an Excel file and leave the "Contacts" menu:



5.1.7.1. Creating contacts.



This button allows you to access to the screen for creating a new contact:

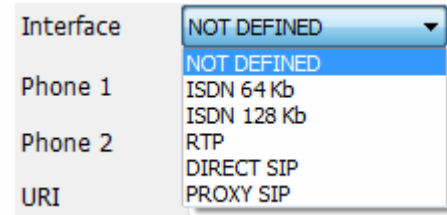


On the left side you can select where to store the new contact (identified with the ✓ symbol). You can store it in the **"General"** agenda and/or in one or several of the codecs defined in the system and that are currently connected.

On the right side you can define the fields for a new contact: name, two ISDN numbers and an URI or IP for network connections. In case the contact is also stored in the **"General"** agenda, you can also define the **"Interface"** field, that is, the type of interface associated to that contact as well as some advanced parameters.

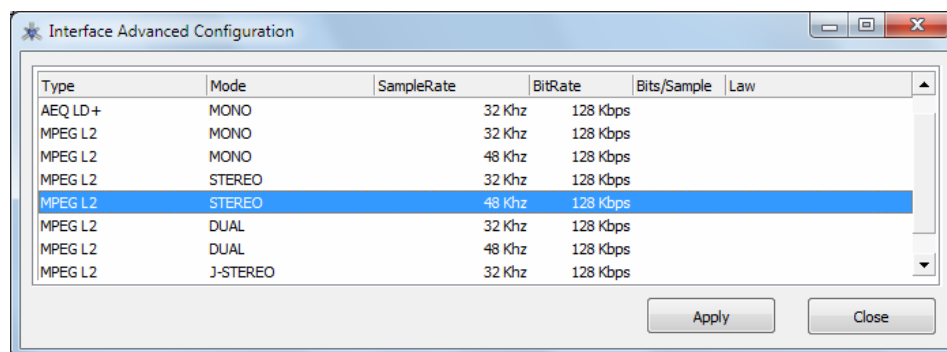
IMPORTANT NOTE: When you create a general agenda contact with a defined "Interface" field (and its advanced parameters) and, in addition, it's also stored in one or several codecs, that additional information is not saved, only the defined "Name", "Phone 1", "Phone 2" and "URI" fields are stored.

Depending on the selected interface, only some fields will be enabled. When "ISDN 64 Kb" is selected only "Phone 1" field can be configured; when "ISDN 128 Kb" is selected only "Phone 1" and "Phone 2" fields are enabled; when "DIRECT SIP" or "PROXY SIP" is selected only "URI" field is enabled (it changes to "IP:Port" when "RTP" is selected).



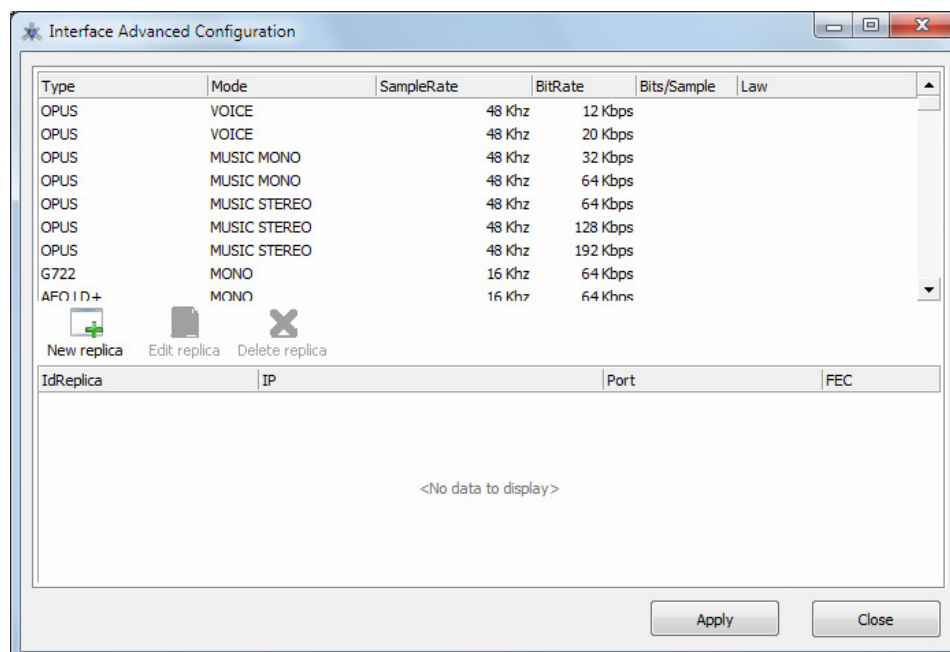
When a interface is selected, the "**Advanced**" button on the right is also enabled. Depending on the selected interface, that button gives access to different advanced configuration parameters:

- When "**ISDN 64 Kb**" or "**ISDN 128 Kb**" is selected you must choose one of the available codings from the list that appears when you press "Advanced":



Type	Mode	SampleRate	BitRate	Bits/Sample	Law
AEQ LD+	MONO	32 Khz	128 Kbps		
MPEG L2	MONO	32 Khz	128 Kbps		
MPEG L2	MONO	48 Khz	128 Kbps		
MPEG L2	STEREO	32 Khz	128 Kbps		
MPEG L2	STEREO	48 Khz	128 Kbps		
MPEG L2	DUAL	32 Khz	128 Kbps		
MPEG L2	DUAL	48 Khz	128 Kbps		
MPEG L2	J-STEREO	32 Khz	128 Kbps		

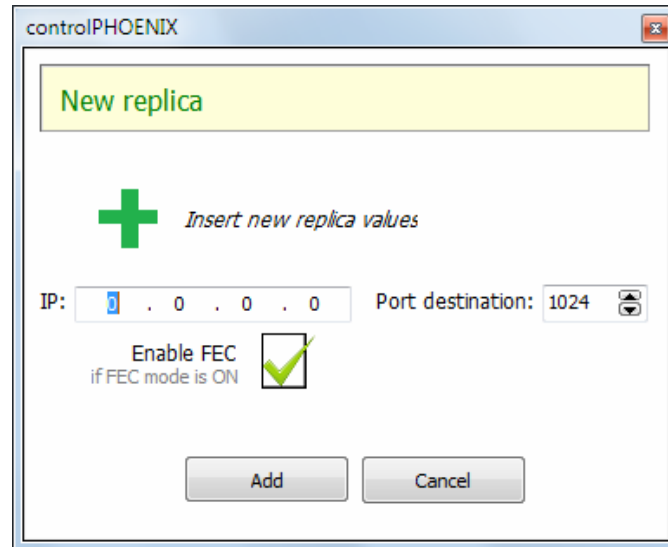
- When "**RTP**" is selected you must choose the encoding mode from the list that appears and, if needed, you can also configure the replicas to send:



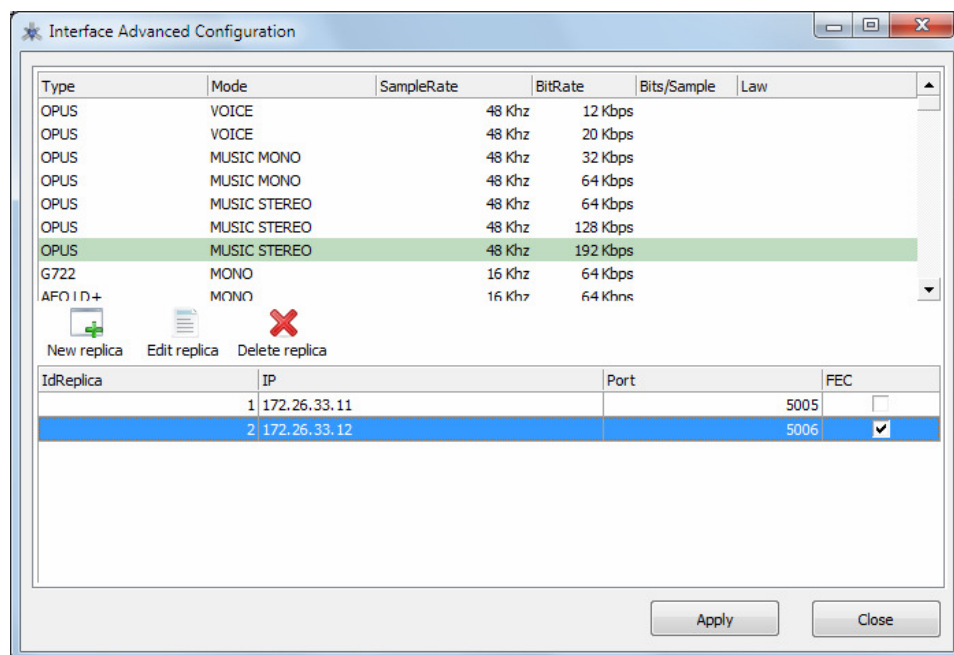
Type	Mode	SampleRate	BitRate	Bits/Sample	Law
OPUS	VOICE	48 Khz	12 Kbps		
OPUS	VOICE	48 Khz	20 Kbps		
OPUS	MUSIC MONO	48 Khz	32 Kbps		
OPUS	MUSIC MONO	48 Khz	64 Kbps		
OPUS	MUSIC STEREO	48 Khz	64 Kbps		
OPUS	MUSIC STEREO	48 Khz	128 Kbps		
OPUS	MUSIC STEREO	48 Khz	192 Kbps		
G722	MONO	16 Khz	64 Kbps		
AEQ LD+	MONO	16 Khz	64 Kbps		

IdReplica	IP	Port	FEC
<No data to display>			

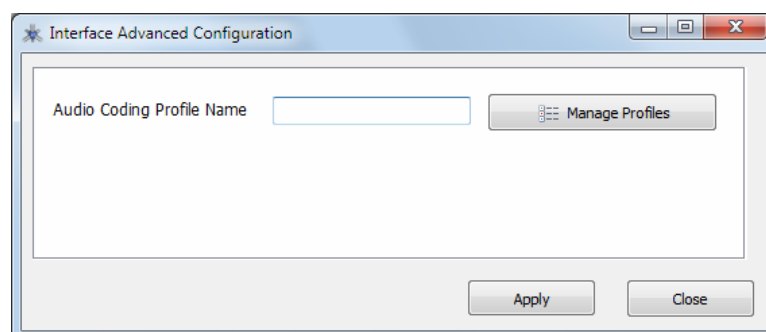
This window allows you to add replicas or parallel streams that are only IP+port pairs we want to send the audio to. Note that FEC can be individually activated or disabled provided that it was enabled (and configured) in the general channel configuration. We can disable it for certain replicas if we want. If the list is empty, the stream will be normally sent to the IP+port indicated in the Call window, as usual. The following window appears when you press the "New replica" button:



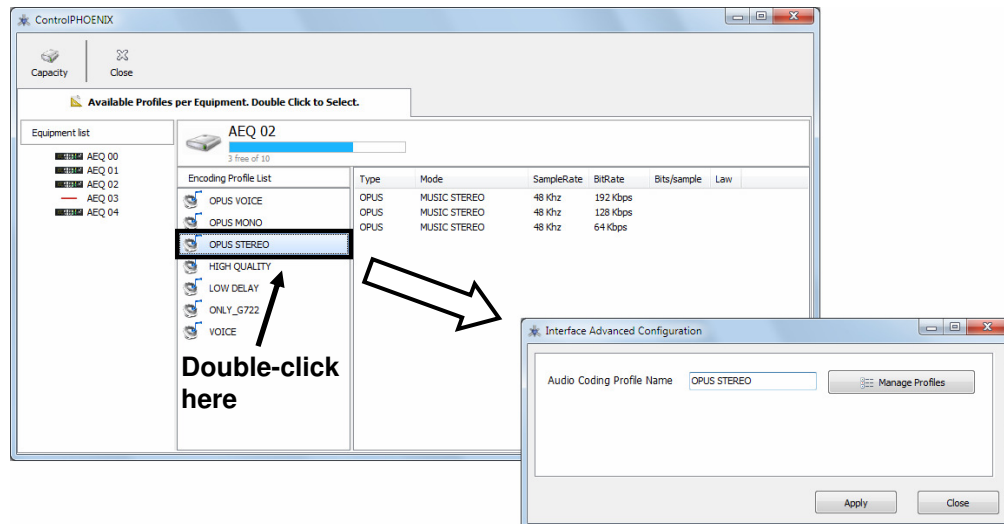
The following window shows a configuration example for an RTP interface transmitting in OPUS MUSIC STEREO 48Khz/192Kbps mode with two replicas added to the main stream (one of them with FEC option activated):



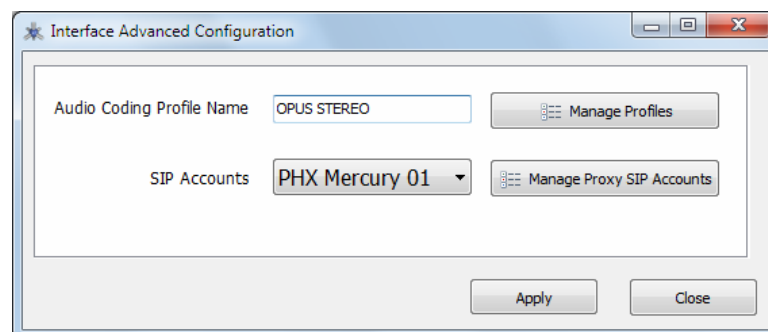
- When "**DIRECT SIP**" is selected you must choose an encoding profile in the window that appears when you press "Advanced":



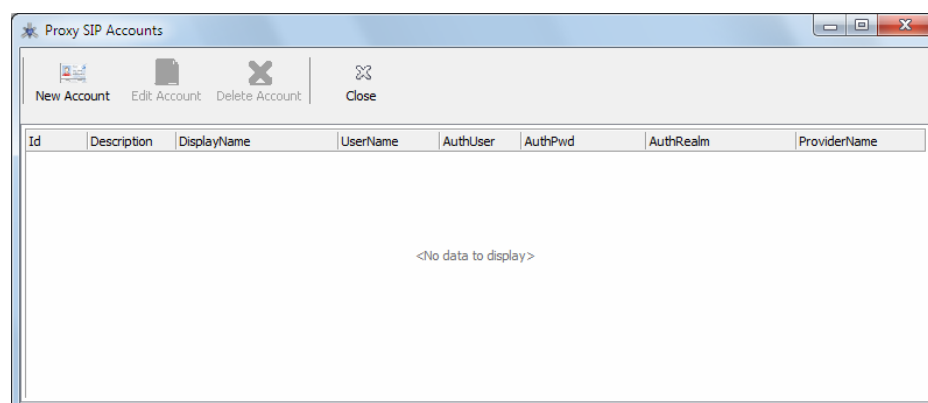
When you call using a "DIRECT SIP" type contact and the profile configured in that contact is not identical to an existing profile of the calling unit, the call will not be made. In order to avoid typing errors when "Audio Coding Profile Name" box is filled, the "Manage Profiles" button gives access to a window that allows you to select a profile from particular equipment and fill that box by double-clicking the desired profile:



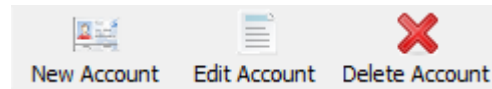
- When "**PROXY SIP**" is selected you must choose an encoding profile (the same way as described previously from "DIRECT SIP") and a SIP server subscription account for the calling unit (this option allows you to use several accounts associated to different SIP servers in the same equipment).



The "Manage Proxy SIP Accounts" button gives access to the window where you can create and manage the SIP accounts:

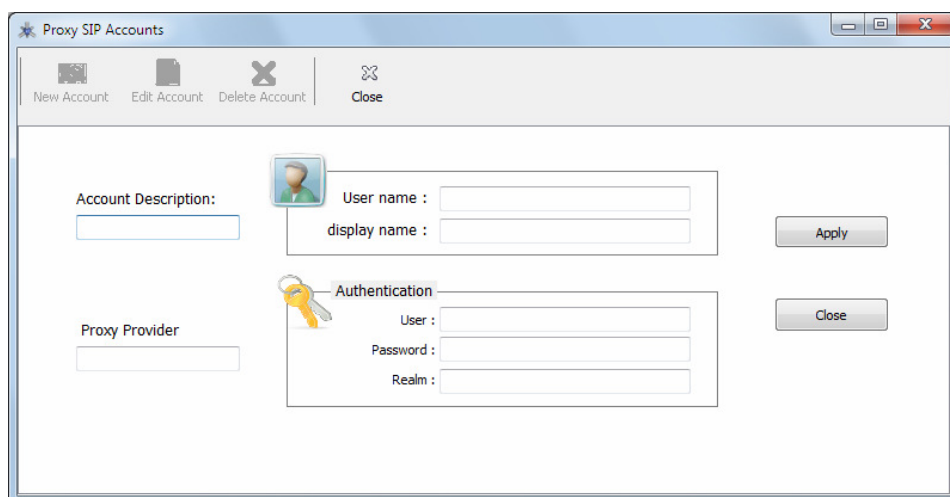


The upper menu includes buttons to add, edit and delete accounts:



When an account is created or edited, a window will appear to configure the parameters associated to SIP protocol:

- **"Account Description"**: account name.
- **"User Name"**: enables you to edit the name of the unit and how it will be reflected in the diverse internal menus of the unit.
- **"Display Name"**: editable name. This is the public name of the equipment that will be used in SIP server, so you can recognize the equipment with this identifier externally to the system.
- **"Proxy Provider"**: enables you to select the external SIP server with which the unit will work (for example, AEQ).
For the moment, it can not be automatically selected (the box must be manually filled) because Providers are configured in each unit and when a contact is created you don't know which calling unit will be used. You should write a valid Provider name for the calling unit when the call is made (it should not be a problem, because only one or two providers are normally used for all the units); in order to check the available Providers for a certain unit, you must select its interface configuration (it should be in Proxy SIP mode), enter "SIP Parameters" and open the "Proxy Provider" drop-down menu or press the "Manage Providers" button (see section 6.1.4.2 of this manual).
- **"Authentication"**: enables you to edit the password and security information for the user profile associated with the unit in the previously selected SIP server:
 - o **"User"**: the "User Name". In most cases, the name of the interface user and the authentication user will be the same (depending on the provider).
 - o **"Pwd"**: the Password associated to that user.
 - o **"Realm"**: the domain where the SIP Server is located, for example sip.aeq.es.



The image shows a window titled "Proxy SIP Accounts". At the top, there is a toolbar with icons for "New Account", "Edit Account", "Delete Account", and a "Close" button. The main area of the window contains several input fields and buttons:

- Account Description:** A text input field.
- User name :** A text input field.
- display name :** A text input field.
- Proxy Provider:** A text input field.
- Authentication:** A section containing:
 - User :** A text input field.
 - Password :** A text input field.
 - Realm :** A text input field.
- Buttons:** An "Apply" button is located to the right of the "User name" and "display name" fields. A "Close" button is located to the right of the "Authentication" section.

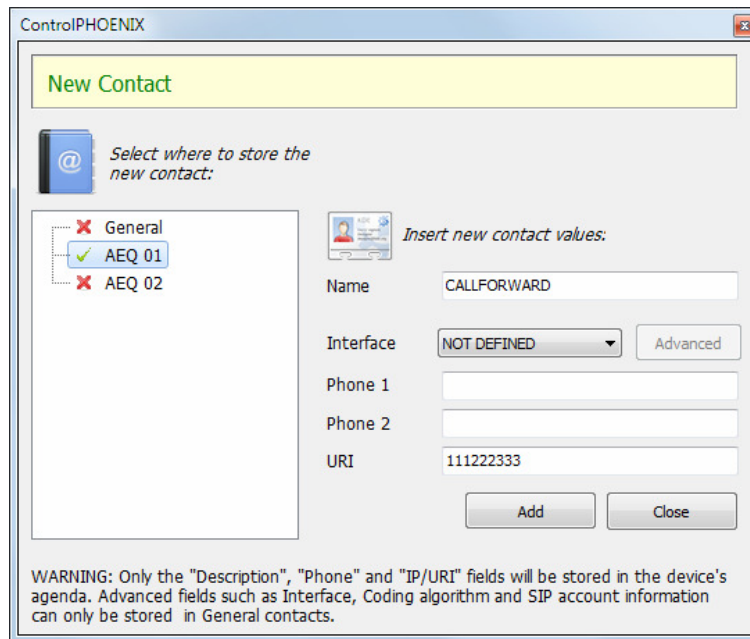
NOTE: You can also access to this option from Tools menu (see section 5.1.8 of this manual). The defined accounts can be used when a new Proxy SIP contact is created, as described before, but also when an equipment is manually configured (see section 6.1.4.2 of this manual).

5.1.7.1.1. CALLFORWARD contact.

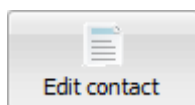
Version **2.2.2.4** (or higher) of **AEQ ControlPHOENIX** application, when acting on Phoenix audiocodescs with CPU version **10.60** or higher, allows call forwarding in PROXY SIP, forwarding the call established at a given time in an audiocodec towards a PHONE: see section 6.7.1 of this manual.

It's possible to make that call forwarding towards a phone number predefined in a contact with a special name: **CALLFORWARD**.

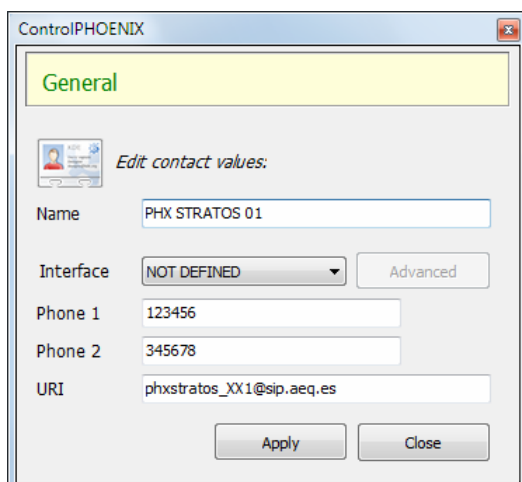
That contact must be created in the unit agenda, not in the "General" one, indicating the phone number or switchboard extension in the "URI" field.



5.1.7.2. Editing contacts.



This buttons allows you to access to the screen for editing an existing contact:



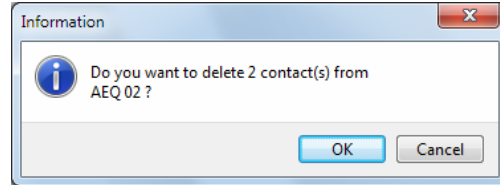
You can edit all the fields of a contact: name, two ISDN numbers and an URI (or IP address). When a contact is edited from the "General" agenda, you can also modify the "Interface" field (and the associated advanced parameters).

NOTE: You can also edit a contact directly in the list of the contacts main screen, by placing in the field you want to modify and making the desired changes. In order to do that the "Contacts Grid Editions" option must be activated (see section 5.1.10.1 of this manual).

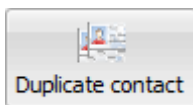
5.1.7.3. Deleting contacts.



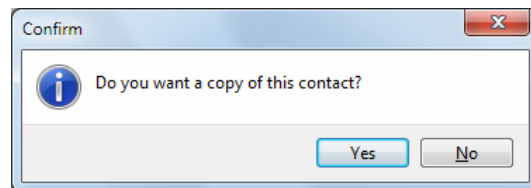
This button allows you to delete one or several selected contacts. Before executing the user will be prompted to confirm the deletion.



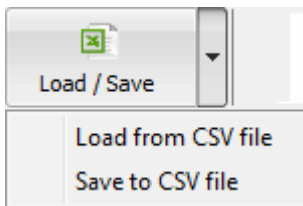
5.1.7.4. Duplicating contacts.



This button allows you to duplicate an existing contact. Before executing the user will be prompted to confirm the duplication.

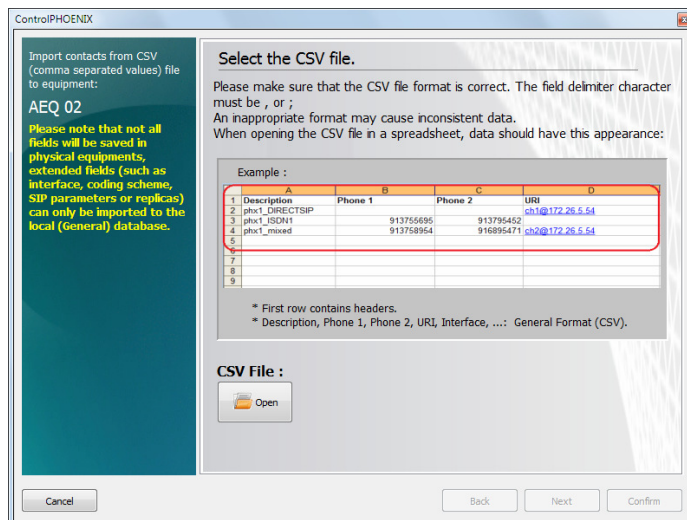
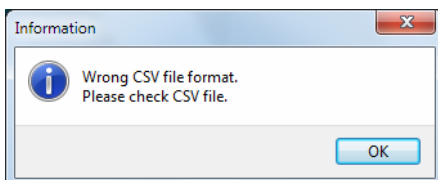


5.1.7.5. Loading/saving contacts list.

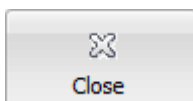


The drop-down menu that appears when you press this button allows you to recover or import the contacts from a CSV file by means of "Load from CSV file" option (the file must have the right format) or to save the contacts to a CSV file by means of "Save to CSV file" option.

When you select "Load from CSV file" option, a screen will show a sample of the format this file must have in order to load it correctly. When you try to load a file with a wrong format an error message will be displayed:



5.1.7.6. Leaving the "Contacts" menu.



This button allows you to leave the "Contacts" menu and returns to the main screen.

5.1.8. Presets.

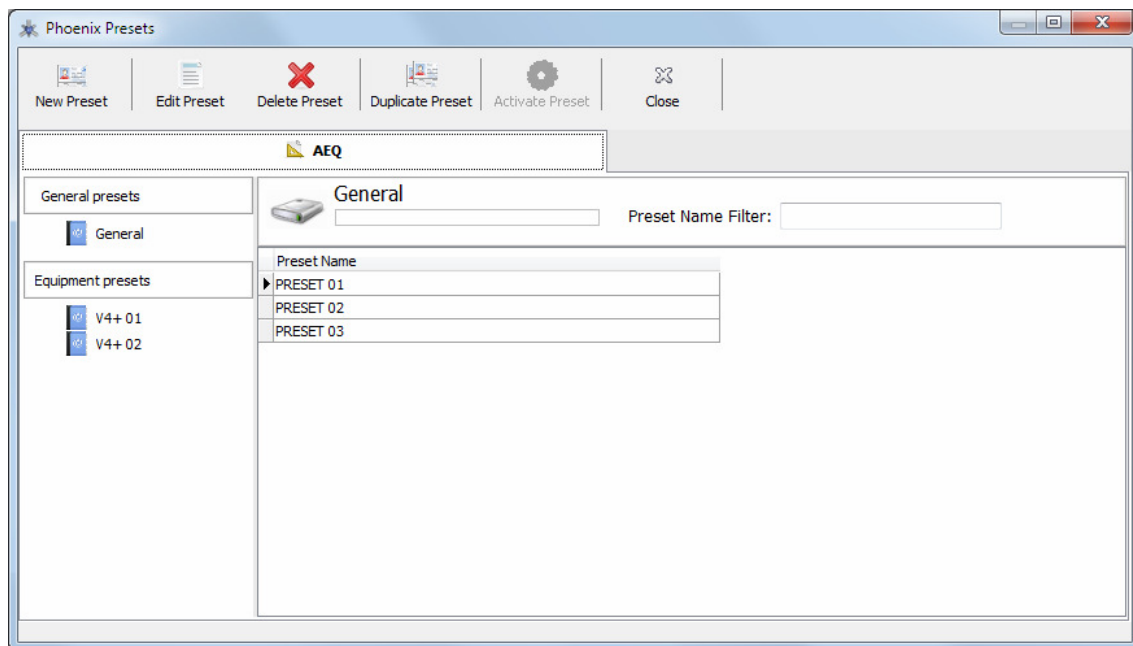


This button provides access to the PRESETs management window. This functionality, available only in audiocodecs with CPU version **10.00** or higher (10.30 in the case of Phoenix ALIO), allows the user to change the configuration of a device, either totally or partially (it can be only some detail), in a very fast way.

PRESETs can be created, edited and saved in the software and sent to the Phoenix VENUS V4, V4+ and ALIO audiocodecs themselves, in order to be called later from the software itself or, in the case of V4+ and ALIO, also from the front panel of the unit.

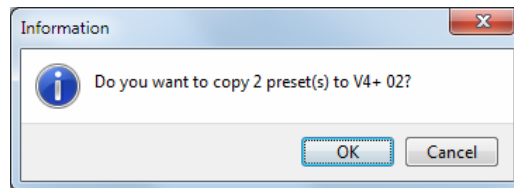
Therefore, it's possible to have a library of PRESETs stored on the PC and thus be able to copy it to the units that we need at any time. This can be useful, among other things, for replicating configuration models to many audiocodecs.

When a PRESET is defined, you will have access to a large part of the possible configuration details in a device, especially regarding the operation of each of its communication channels 1 and 2 (if it has both) independently. Regarding the common part, the outgoing remote control setting can be changed.



- **General presets:** the upper left part of the screen shows the general list of PRESETs, that is, the one stored in the PC where the application is installed.
- **Equipment presets:** the left part of this area shows also below the access to the individual PRESETs of the units connected to the system, that is, the list of presets stored into each equipment internal non-volatile RAM.
- **Presets detail:** at the right part you can see a list of the presets associated to the general list or to the selected codec. On top you can see the "General" indication or the name of the selected codec and, in that case, the number of available presets (of a total of 10 for each unit). The "Preset Name Filter" box allows you to filter the list by the characters that appears in "Preset Name" field. Below that, you can see the presets list; one line is for each preset, where appears the "Preset Name".

You can **copy PRESETs between lists** through simple "drag and drop". You must select the PRESETs to be copied from the general list or a codec list (on the right part of the screen) and drag and drop it to the desired codec or the general list (on the left part of the screen). A confirmation window will appear:



The upper menu of the main window includes buttons to add, edit, delete and duplicate PRESETs, activate a PRESET on a certain device and leave the "Presets" menu:

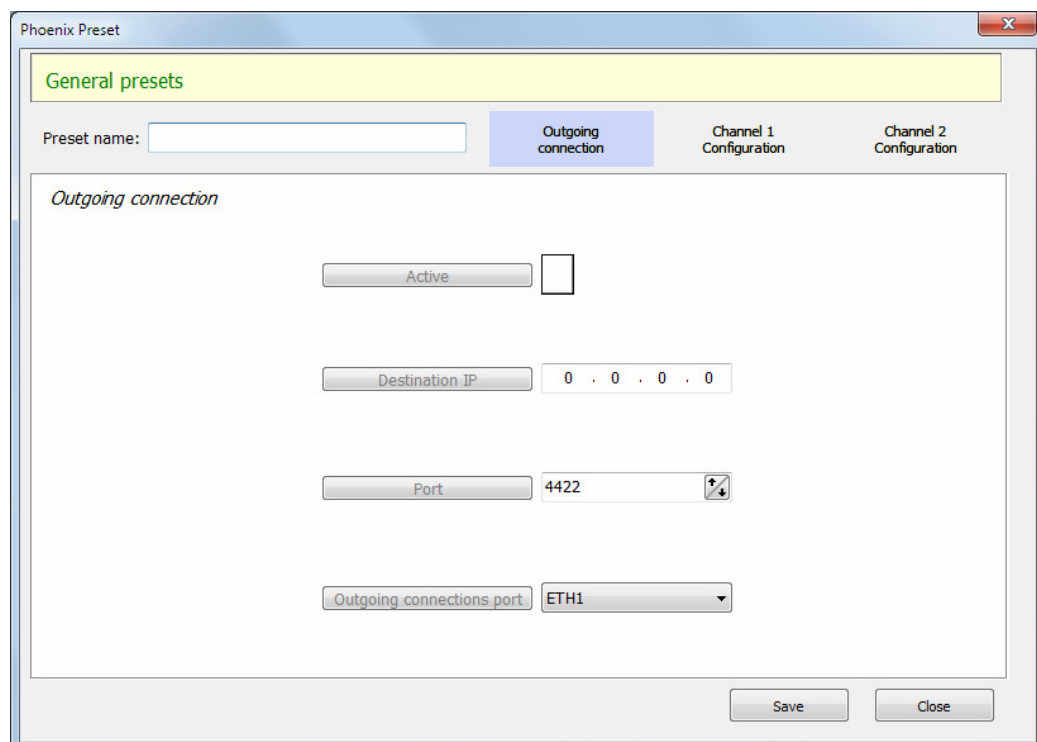


5.1.8.1. Creating presets.



When a type of list is selected (General or that of a particular device) and the "New Preset" button is pressed, a window with three tabs appears. By default, when creating a new PRESET, all the fields are deactivated (if applied, nothing will be modified); it's necessary to activate the button associated to some field so that this PRESET has some effect when activated in a device.

- **Device general settings:** The only general setting that applies to the entire device that can be changed by means of a PRESET is the one related to the outgoing remote control connection ("**Outgoing connection**" tab):



As you can see, there are 4 parameters that can be configured:

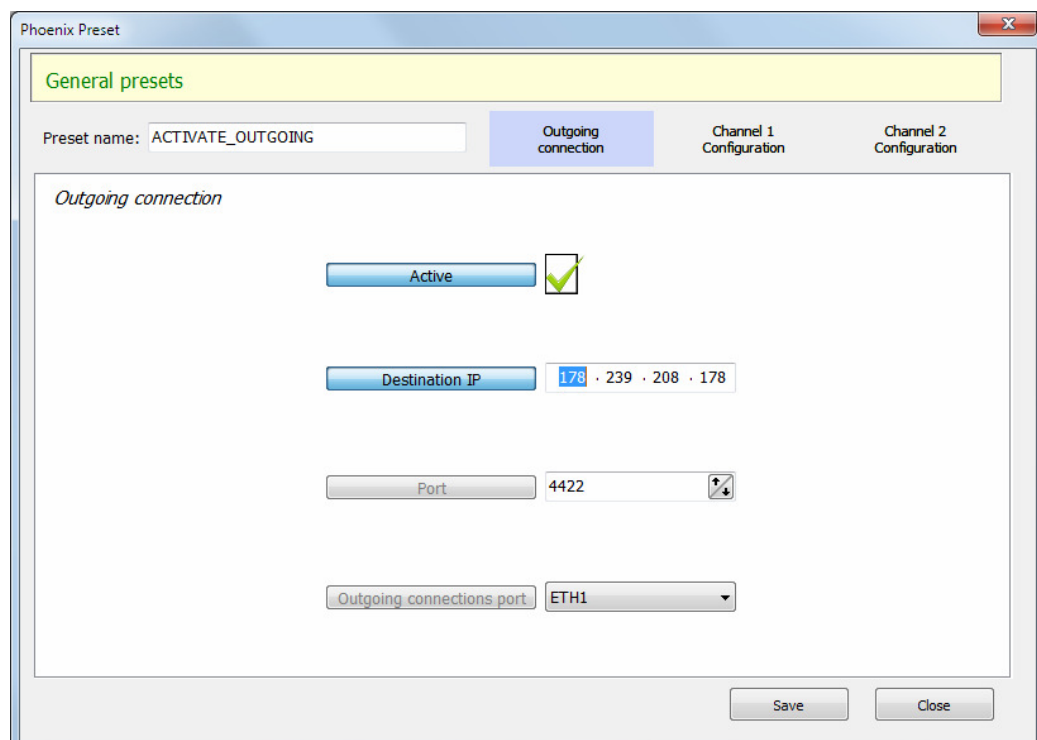
- **"Active"**: allows you to configure whether the outgoing control connection is active or not.
- **"Destination IP"**: public IP address of the studio.
- **"Port"**: port used for control.
- **"Outgoing connection port"**: physical Ethernet interface of the audiocodec used for the connection.

Note that each of the parameters has a default value, and a button at the left with its name, which appears in **gray** if this particular parameter will not be affected by the loading of the PRESET being edited, or **highlighted** if the specified value for that parameter will be CHANGED in the device when that PRESET is loaded. It works like this for any of the configurable parameters in the PRESETs and allows very specific changes using a PRESET in which only one parameter has been marked, or an almost complete reconfiguration activating and defining many parameters.

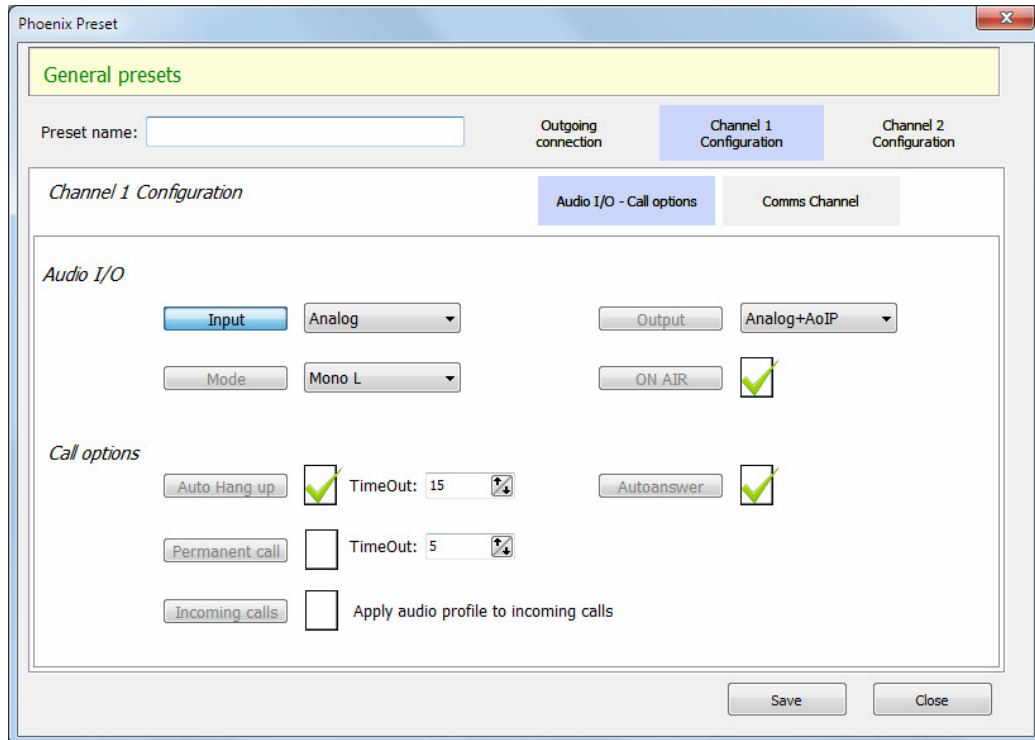
If any of the parameters is modified, the button at the left (the one that indicates the name of the parameter) will be automatically activated, indicating that when this PRESET is loaded it will be modified with the indicated value.

NOTE: If a parameter is changed accidentally (or just for testing), its label will be activated. If you really don't want that parameter to change, you have to click on the label again so that parameter is not modified when the PRESET is loaded.

For example, if you simply want to activate remote control on a device and define the IP address of your studio, without changing the port number used or the interface, the screen would look like this:



- **Settings for each channel:** The "Channel 1 Configuration" and "Channel 2 Configuration" tabs allow you to configure channels 1 and 2 of the audiocodec and consist of 2 tabs: "Audio I/O - Call options" and "Comms Channel".



Phoenix Preset

General presets

Preset name:

Outgoing connection | **Channel 1 Configuration** | Channel 2 Configuration

Channel 1 Configuration

Audio I/O - Call options | Comms Channel

Audio I/O

Input: Analog | Output: Analog+AoIP

Mode: Mono L | ON AIR: ☒

Call options

Auto Hang up: ☒ TimeOut: 15 | Autoanswer: ☒

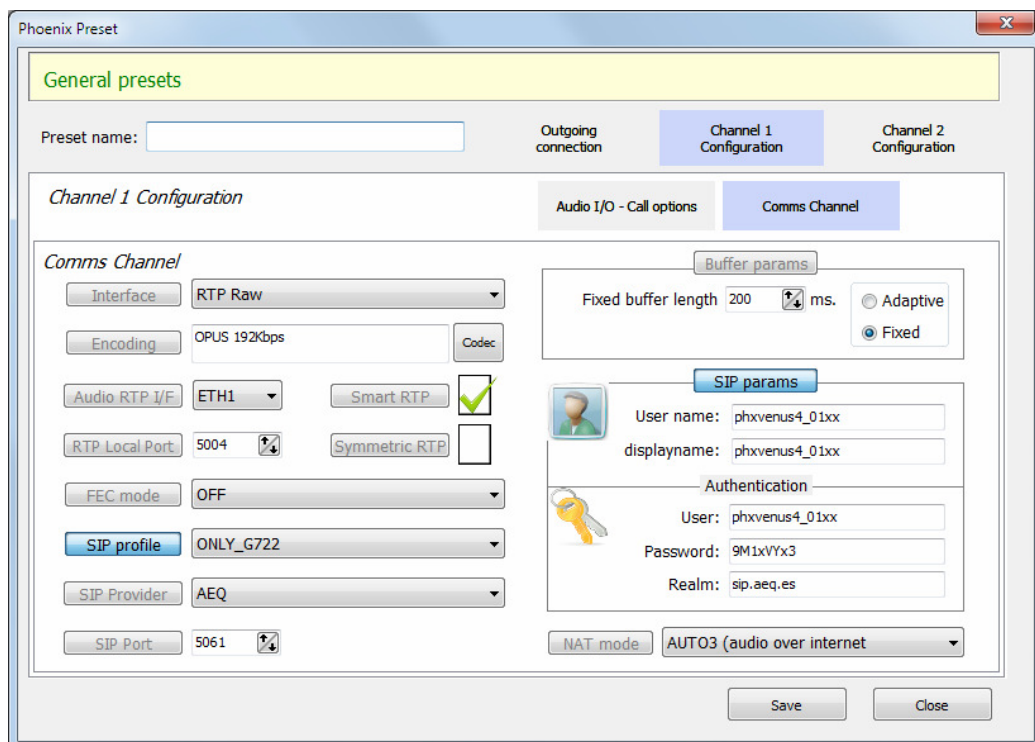
Permanent call: ☐ TimeOut: 5

Incoming calls: ☐ Apply audio profile to incoming calls

Save | Close

As shown in the previous image, in the first of these tabs you can configure the audio inputs/outputs ("Audio I/O", at the top), as well as various call options on that channel ("Call options").

In the next tab, "Comms Channel", it's possible to select the communication mode, as well as the interface used for RTP, encoding (for RTP), profiles (for SIP), port number, FEC, buffer settings and credential parameters of SIP.



Phoenix Preset

General presets

Preset name:

Outgoing connection | **Channel 1 Configuration** | Channel 2 Configuration

Channel 1 Configuration

Audio I/O - Call options | **Comms Channel**

Comms Channel

Interface: RTP Raw

Encoding: OPUS 192Kbps | Codec

Audio RTP I/F: ETH1 | Smart RTP: ☒

RTP Local Port: 5004 | Symmetric RTP: ☐

FEC mode: OFF

SIP profile: ONLY_G722

SIP Provider: AEQ

SIP Port: 5061

Buffer params

Fixed buffer length: 200 ms. | Adaptive: ☐ Fixed: ☒

SIP params

User name: phxvenus4_01xx

displayname: phxvenus4_01xx

Authentication

User: phxvenus4_01xx

Password: 9M1xVYx3

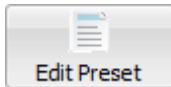
Realm: sip.aeq.es

NAT mode: AUTO3 (audio over internet)

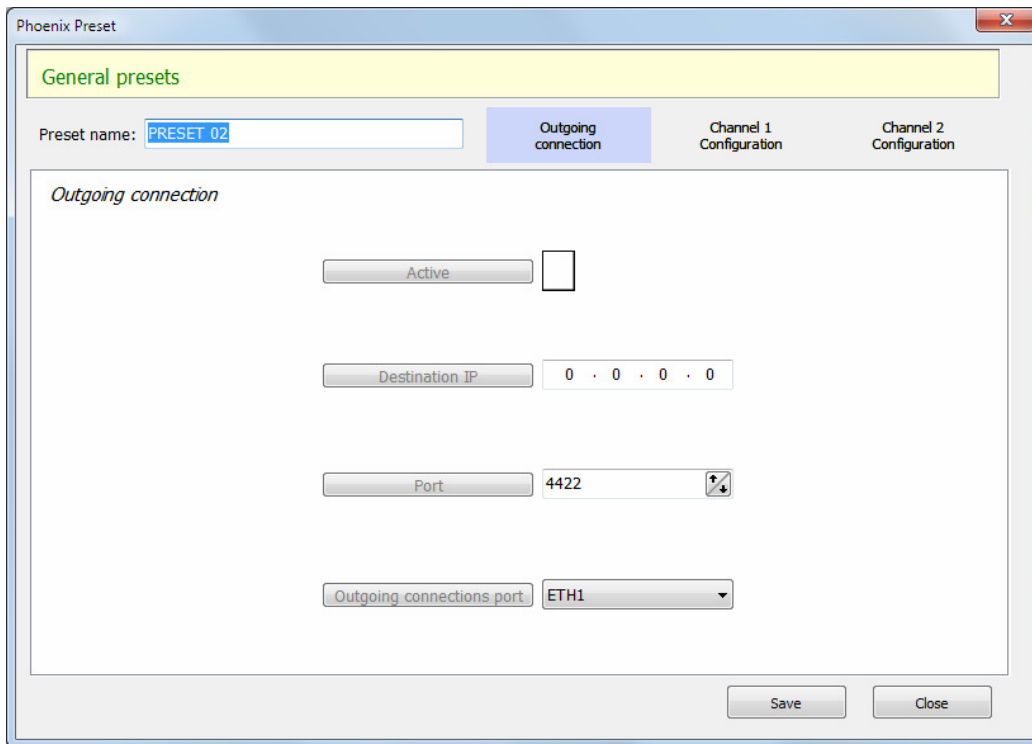
Save | Close

Of course, these two tabs also appear for channel 2, and it's possible to define whether or not to modify each parameter, independently.

5.1.8.2. Editing presets.



This buttons allows you to access to the screen for editing a PRESET of the list:



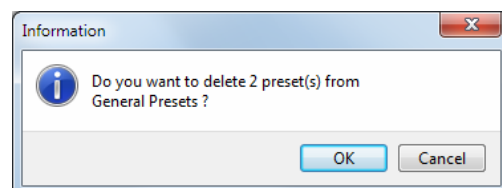
The Phoenix Preset dialog box is titled "Phoenix Preset" and has a close button (X) in the top right corner. It features a tabbed interface with three tabs: "General presets" (selected), "Outgoing connection", "Channel 1 Configuration", and "Channel 2 Configuration". Under the "General presets" tab, there is a "Preset name:" label followed by a text field containing "PRESET 02". Below this, the "Outgoing connection" section is visible, containing several fields: an "Active" checkbox, a "Destination IP" field with the value "0 . 0 . 0 . 0", a "Port" field with the value "4422" and a small up/down arrow icon, and an "Outgoing connections port" dropdown menu currently set to "ETH1". At the bottom right of the dialog are "Save" and "Close" buttons.

You can edit all the fields of a PRESET.

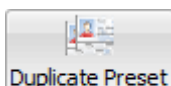
5.1.8.3. Deleting presets.



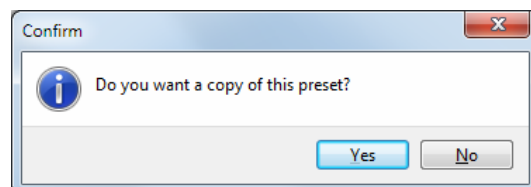
This button allows you to delete one or several PRESETS. Before executing the user will be prompted to confirm the deletion.



5.1.8.4. Duplicating presets.



This button allows you to duplicate a PRESET. Before executing the user will be prompted to confirm the duplication.



5.1.8.5. Activation of presets.

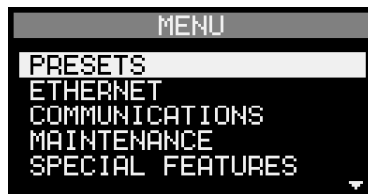
Once the desired PRESET is copied to a device (this action has no effect on its configuration, it's only saved in its internal non-volatile memory for future use), it's possible to activate it at any time. There are several ways to activate a PRESET.

- 1) On the front panel of Phoenix Venus V4+ there is a dedicated key at the right of the display:

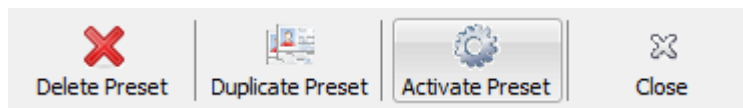


By pressing this key, the list of PRESETs saved in the device will appear and one of them can be selected by turning the "SEL" encoder and confirm by pressing it. Confirmation is requested.

In the case of Phoenix ALIO, the "MENU" key takes you to the menu where the "PRESETS" option gives access to the list saved in the device. By turning and pressing the "NAVI / Ch" encoder you can select and load any of the presets of the list.

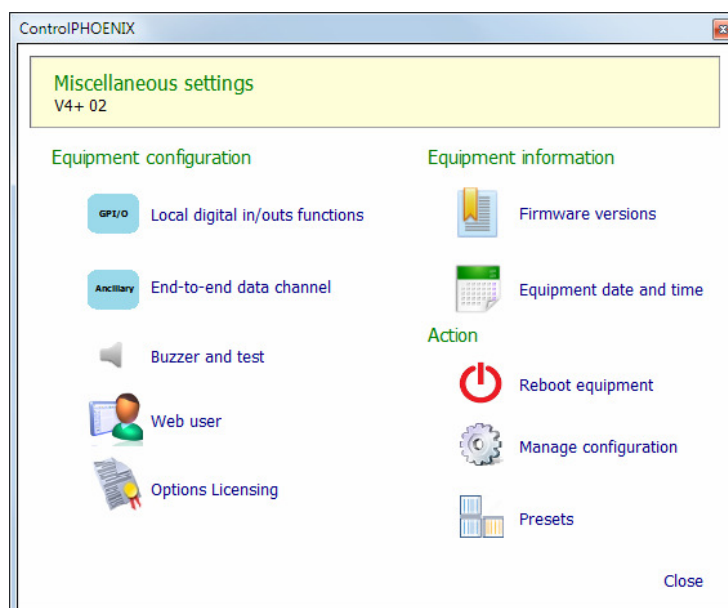


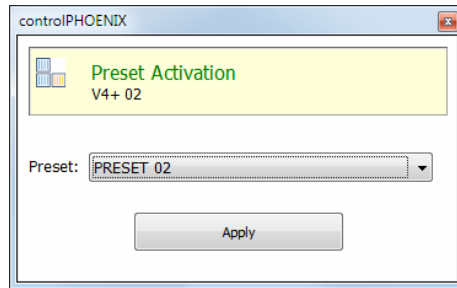
- 2) From the software, in the PRESETs configuration list itself, it's possible to select a PRESET in a particular codec and then click on the "**Activate Preset**" button on the top bar:



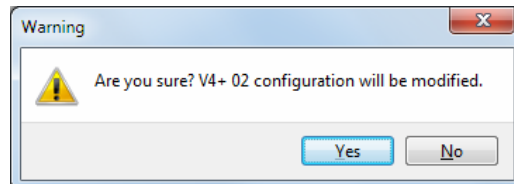
Note that this button will appear disabled unless a PRESET saved on one of the devices is selected.

- 3) Also from the software, by pressing on "**CONFIGURATION**" → "**Miscellaneous**", by clicking on the "Presets" option you access the list of PRESETs saved on the unit and you can choose the one you want:



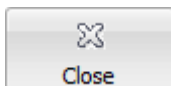


Once the PRESET is selected, press "Apply": confirmation is requested.



If you don't wish to activate the PRESET, it's possible to deny this activation and then close the previous window using the upper right cross.

5.1.8.6. Leaving the "Presets" menu.

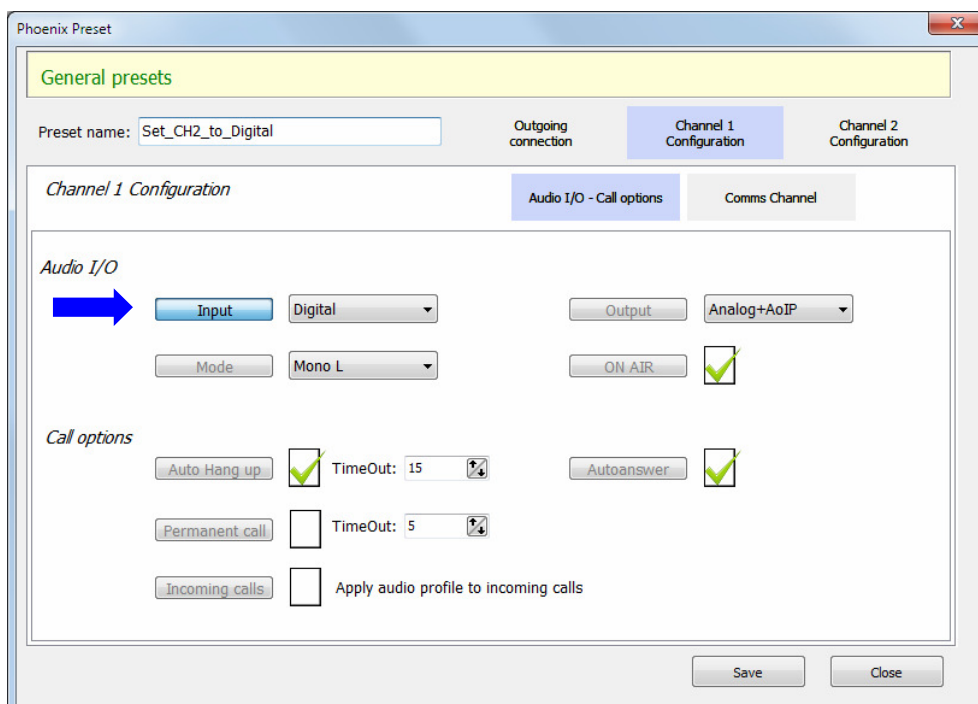


This button allows you to leave the "Presets" menu and returns to the main screen.

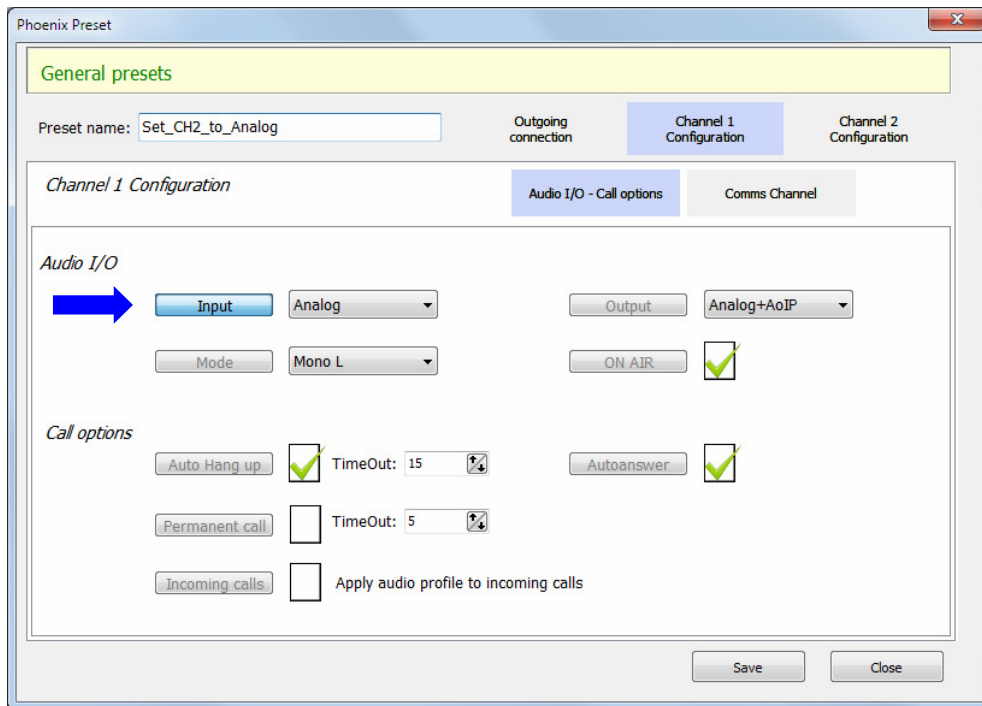
5.1.8.7. Presets examples for Phoenix Venus V4 / V4+.

Example 1: Changing the input mode of channel 2 to analog using PRESET.

If you simply want to execute a PRESET (which we will call "Set_CH2_to_Digital") that changes the input mode from analog to digital only on channel 2, you will have to create a PRESET in which only the "Channel 2 Configuration" → "Audio I/O - Call options" → "Audio I/O" → "Input" option is active, and the "Digital" mode will be selected.



The logical thing would be to create another PRESET (which we will call "Set_CH2_to_Analog") that changes the input mode of channel 2 back to analog. We can start from the same PRESET, duplicating it, then changing its name and, finally, the particular value of that field, which will now be "Analog":

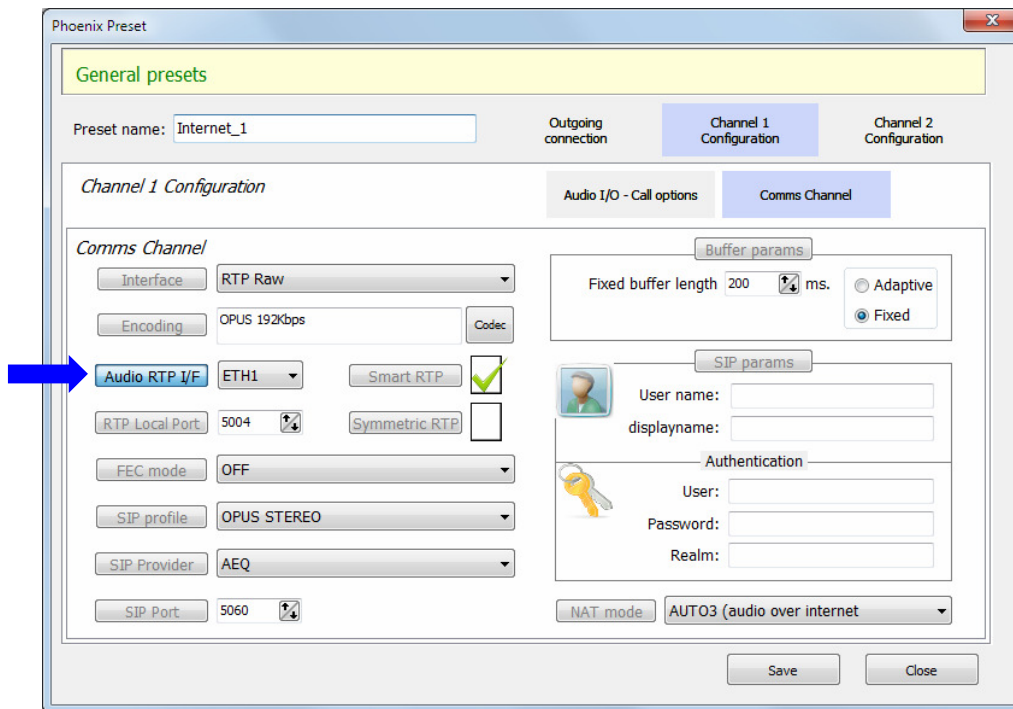


In operation, one or the other PRESET will be executed to switch the input mode of channel 2 between analog and digital. We can do it from the software or call it by pressing the "PRESET" key on the front panel (in the case of Phoenix Venus V4+) and selecting the item "Set_CH2_to_Analog" from the list.

Example 2a: Internet provider redundancy in RTP using PRESET.

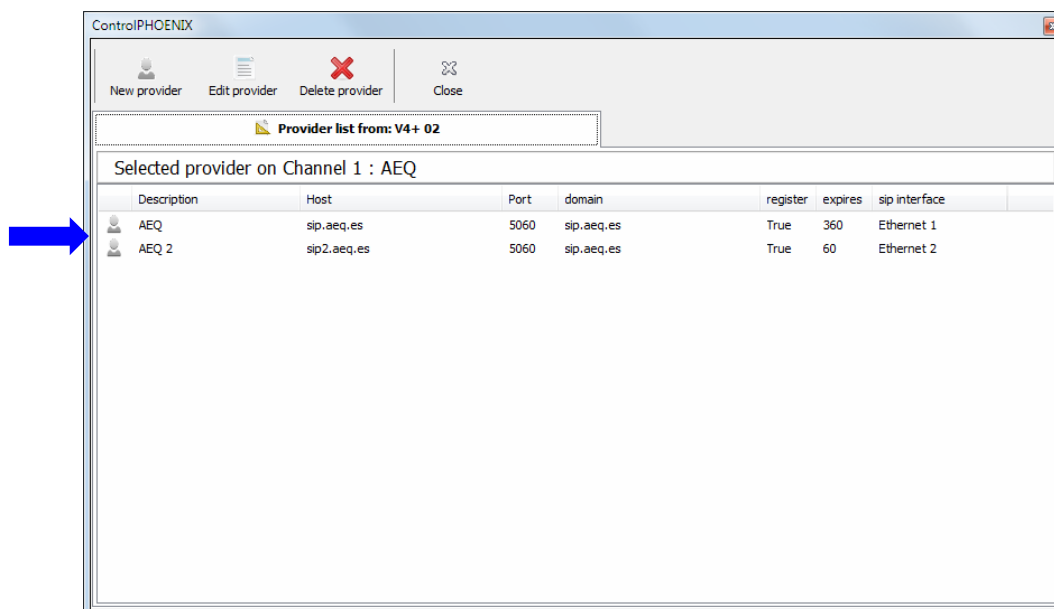
Let's suppose we have Internet access from a different provider connected to each NET 1 and NET2 port, one main and one backup. To easily switch communication from one to the other in case of main access failure, we can do the following.

By means of a PRESET it's possible to change the RTP interface for audio to ETH1 or ETH2, in a similar way to how it was explained to change the input mode in example 1. If the device is the one launching the call, there will be no problem, since we will call the same destination IP and port. If the device is the one receiving the call, it will have a different public IP and port, so the remote end must know about this change.

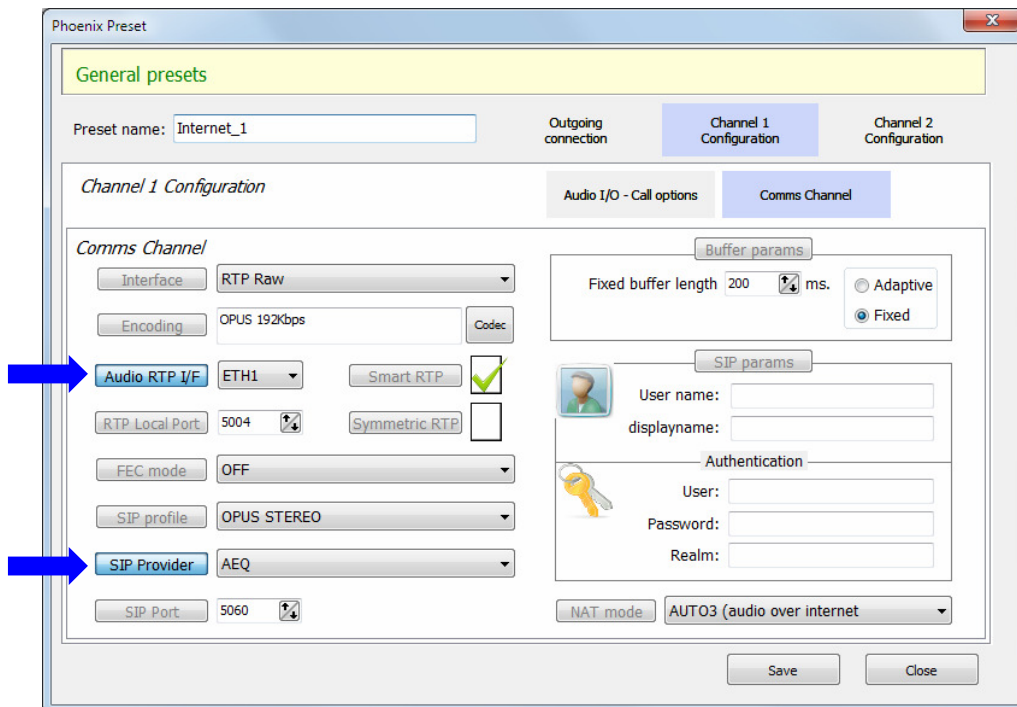


Example 2b: Internet provider redundancy in SIP using PRESET.

As explained in example 2a, if you want to implement Internet provider redundancy, connecting a different access to each port, but you are working in SIP mode, you can use the same account on the same server, but you will have to create 2 providers ("AEQ" and "AEQ 2", in the following example), which really have the same host, port and domain, but in one it's defined as "SIP Interface" Ethernet 1 and in the other one Ethernet 2:

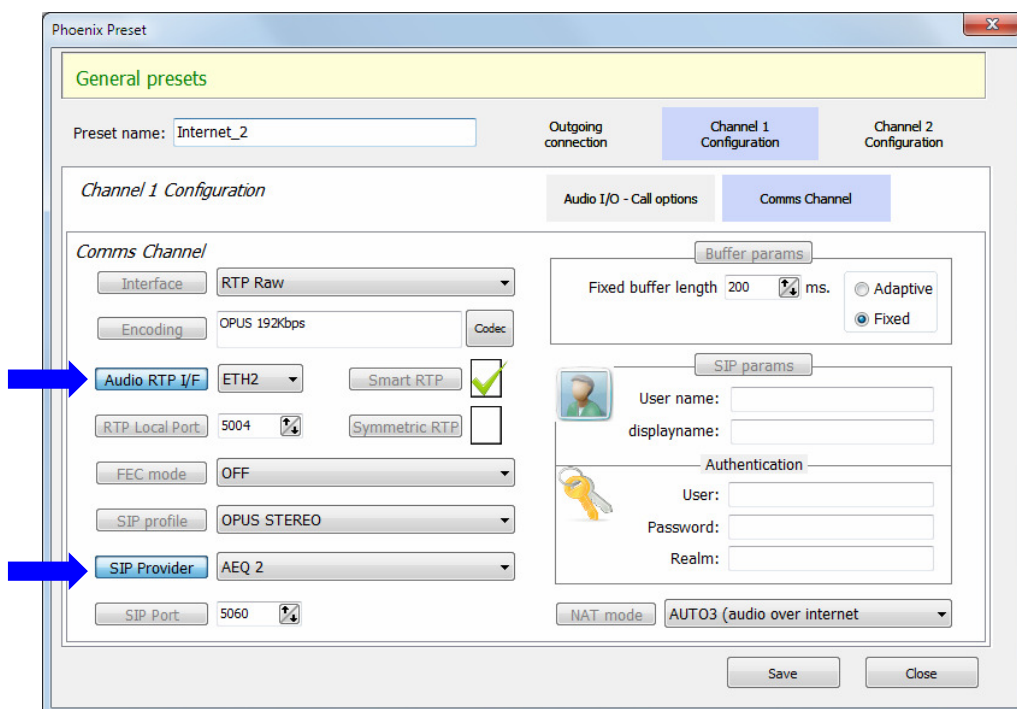


Of course, for the audio to work, it's also necessary to change the RTP interface for audio to the corresponding connector on the corresponding channel (or channels), as explained in example 2a.



The screenshot shows the 'Phoenix Preset' window with the 'General presets' tab selected. The 'Preset name' is 'Internet_1'. The 'Channel 1 Configuration' tab is active. Under 'Comms Channel', the 'Audio RTP I/F' is set to 'ETH1' and the 'SIP Provider' is set to 'AEQ'. Blue arrows point to these two settings. Other visible settings include 'Interface: RTP Raw', 'Encoding: OPUS 192Kbps', 'RTP Local Port: 5004', 'FEC mode: OFF', 'SIP profile: OPUS STEREO', 'SIP Port: 5060', 'Buffer params: Fixed 200 ms', and 'NAT mode: AUTO3 (audio over internet)'.

Preset "Internet_1" to change to the provider connected to NET1 (shown on CH1)



The screenshot shows the 'Phoenix Preset' window with the 'General presets' tab selected. The 'Preset name' is 'Internet_2'. The 'Channel 1 Configuration' tab is active. Under 'Comms Channel', the 'Audio RTP I/F' is set to 'ETH2' and the 'SIP Provider' is set to 'AEQ 2'. Blue arrows point to these two settings. Other visible settings include 'Interface: RTP Raw', 'Encoding: OPUS 192Kbps', 'RTP Local Port: 5004', 'FEC mode: OFF', 'SIP profile: OPUS STEREO', 'SIP Port: 5060', 'Buffer params: Fixed 200 ms', and 'NAT mode: AUTO3 (audio over internet)'.

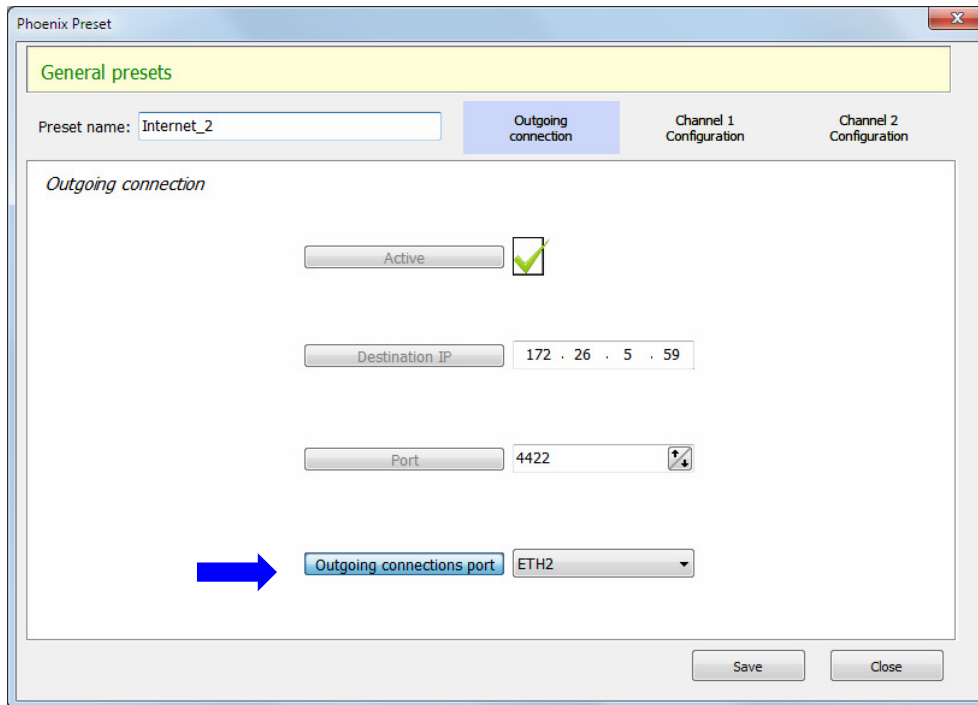
Preset "Internet_2" to change to the provider connected to NET2 (shown on CH1)

This must be done on the corresponding channel (or channels). The previous examples show the edition of the preset for CH1, but if you want to use CH2 as well, you should make the same changes to that channel.

Switching from one provider to another may require a few seconds until the SIP server takes note of the new IP that is registered for that user and routes the calls correctly.

Example 2c: Remote control redundancy using PRESET.

Continuing with the previous case, in which we have 2 Internet accesses, one on each NET1 and NET2 port, in order to keep the outgoing remote control operational (the codec actively connects to the studio's public IP), it will be necessary to incorporate to the presets described above, a change of the port (ETH1 / ETH2) through which outgoing control connections are sent to recover control of the device:



Phoenix Preset

General presets

Preset name:

Outgoing connection | Channel 1 Configuration | Channel 2 Configuration

Outgoing connection

Active ☒

Destination IP

Port

Outgoing connections port

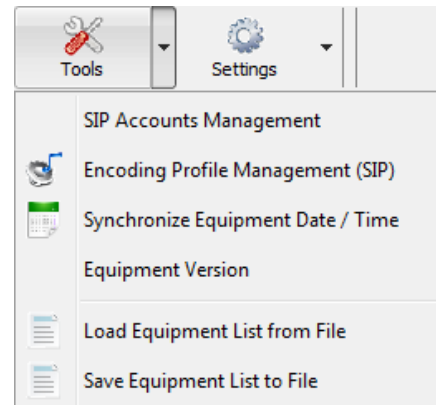
Save Close

Preset to change the outgoing control connection to the Internet provider connected to NET2

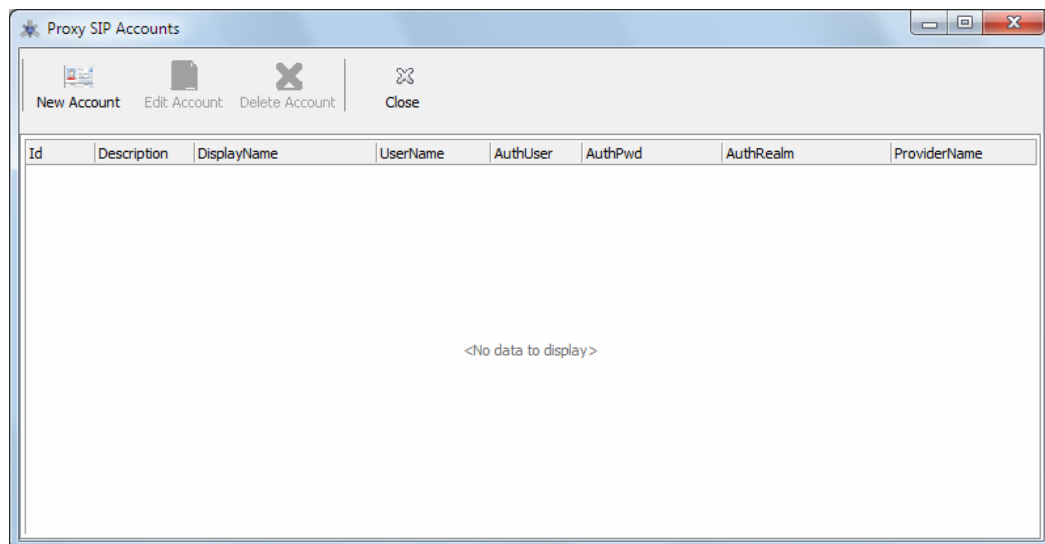
5.1.9. Tools.

This menu includes several tools. The most important one is **"Encoding Profile Management (SIP)"**, that allows to manage the encoding profiles used in SIP based IP connections.

The 6 available tools are described below.



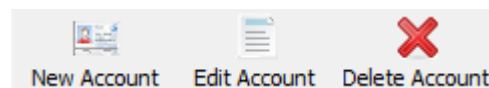
- "SIP Accounts Management".



From this window you can create and manage the SIP accounts.

NOTE: You can also access to this option from the window where Proxy SIP type contacts are created/edited (see section 5.1.7.1 of this manual) or when equipment is manually configured (see section 6.1.4.2 of this manual).

The upper menu includes buttons to add, edit and delete accounts:

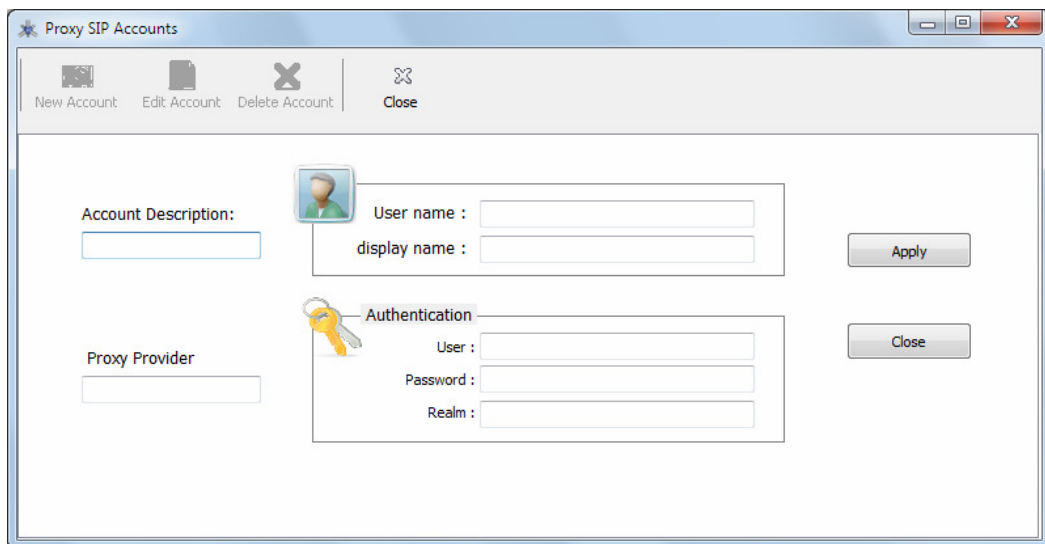


When an account is created or edited, a window will appear to configure the parameters associated to SIP protocol:

- **"Account Description"**: account name.
- **"User Name"**: enables you to edit the name of the unit and how it will be reflected in the diverse internal menus of the unit.
- **"Display Name"**: editable name. This is the public name of the equipment that will be used in SIP server, so you can recognize the equipment with this identifier externally to the system.
- **"Proxy Provider"**: enables you to select the external SIP server with which the unit will work (for example, AEQ).

For the moment, it can not be automatically selected (the box must be manually filled) because Providers are configured in each unit and when a contact is created you don't know which calling unit will be used. You should write a valid Provider name for the calling unit when the call is made (it should not be a problem, because only one or two providers are normally used for all the units); in order to check the available Providers for a certain unit, you must select its interface configuration (it should be in Proxy SIP mode), enter "SIP Parameters" and open the "Proxy Provider" drop-down menu or press the "Manage Providers" button (see section 6.1.4.2 of this manual).

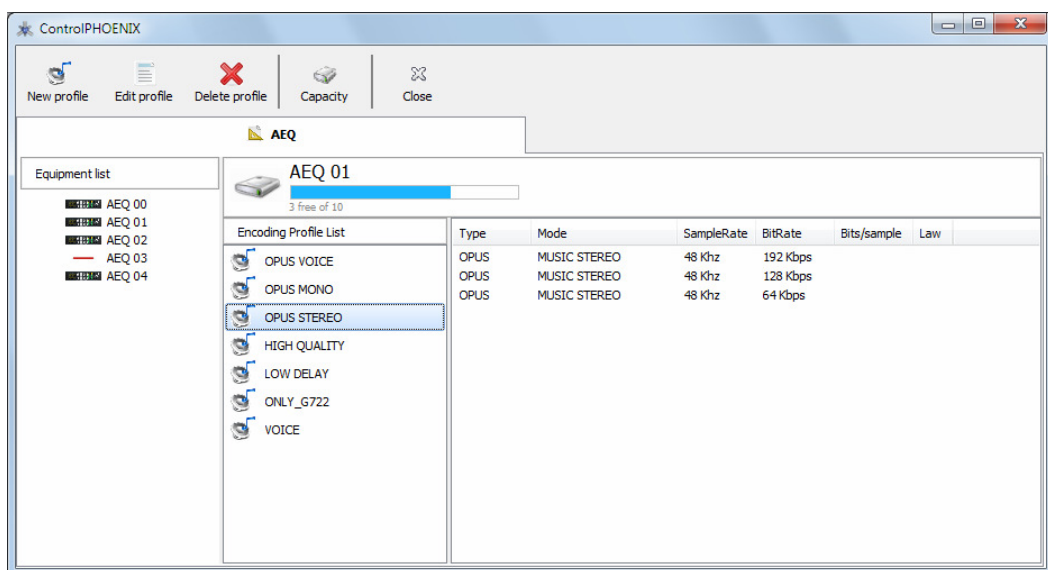
- **"Authentication"**: enables you to edit the password and security information for the user profile associated with the unit in the previously selected SIP server:
 - o **"User"**: the "User Name". In most cases, the name of the interface user and the authentication user will be the same (depending on the provider).
 - o **"Pwd"**: the Password associated to that user.
 - o **"Realm"**: the domain where the SIP Server is located, for example sip.aeq.es.



The "Proxy SIP Accounts" window contains the following fields and buttons:

- Buttons:** New Account, Edit Account, Delete Account, Close.
- Account Description:** A text input field.
- User name:** A text input field.
- display name:** A text input field.
- Proxy Provider:** A text input field.
- Authentication:** A section containing:
 - User:** A text input field.
 - Password:** A text input field.
 - Realm:** A text input field.
- Buttons:** Apply, Close.

- **"Encoding Profile Management (SIP)".**



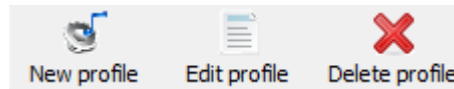
The "ControlPHOENIX" window displays the following information:

- Buttons:** New profile, Edit profile, Delete profile, Capacity, Close.
- Equipment list:**
 - AEQ 00
 - AEQ 01
 - AEQ 02
 - AEQ 03
 - AEQ 04
- AEQ 01:**
 - 3 free of 10
 - Encoding Profile List:**
 - OPUS VOICE
 - OPUS MONO
 - OPUS STEREO** (selected)
 - HIGH QUALITY
 - LOW DELAY
 - ONLY_G722
 - VOICE
 - Table:**

Type	Mode	SampleRate	BitRate	Bits/sample	Law
OPUS	MUSIC STEREO	48 Khz	192 Kbps		
OPUS	MUSIC STEREO	48 Khz	128 Kbps		
OPUS	MUSIC STEREO	48 Khz	64 Kbps		

This option allows you to manage the encoding profiles for SIP connections.

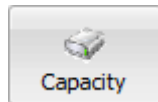
The upper menu includes buttons to add, edit and delete profiles:



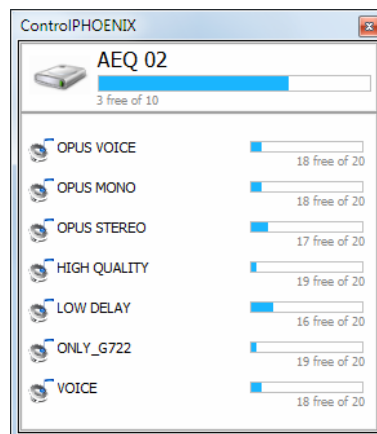
Basically, a profile is a list of one or more defined audio coding formats that will be used by SIP protocol in order to negotiate the particular coding to be used during the IP call. When an equipment calls another using SIP, this profile is sent to the called unit so it can select among several possibilities (as it may not support all of the coding formats the calling equipment supports).

So the user can select among one or other list (profile) for each call (see section 6.1.3.1). The order within the list is important, as priority will be assigned from top to bottom, and in fact the most usual situation is that the called equipment accepts the first coding in the profile.

When a new profile is created, it will be created in the selected equipment at that moment. Once created and configured, the profile can be copied to other equipment using drag and drop Windows procedure.

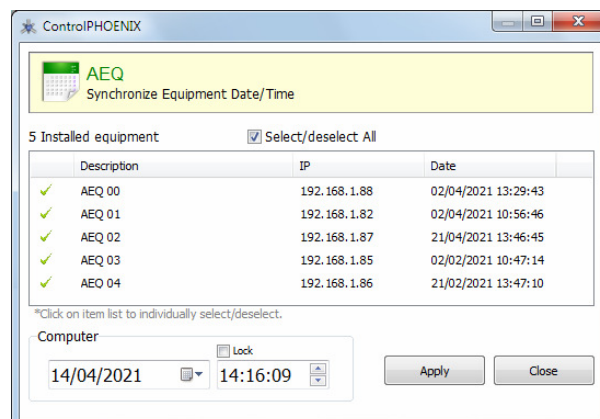


The **Capacity** button allows you to access to a window where the profiles list associated to a certain unit is shown, as well as the number of free positions for new profiles (up to 10 per unit) and the number of free positions inside each profile for adding new coding formats (up to 20 per profile).



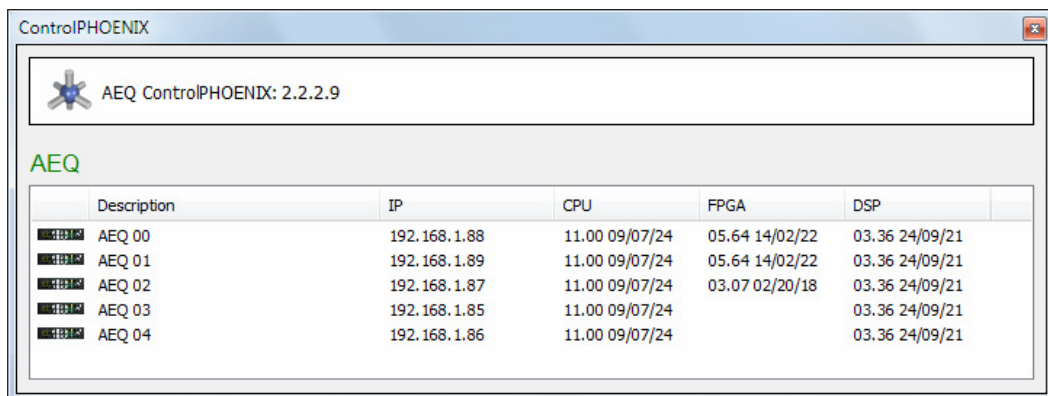
- "Synchronize Equipment Date / Time".

This screen allows us to synchronize the codecs and keep them in sync with the computer's date and time.



- "Equipment Version".

This option shows on a window a list of the IP address for each codec and displays the firmware versions of each of the programmable elements inside it (CPU, FPGA and DSP for Phoenix ALIO, Phoenix Stratos and Phoenix Studio; CPU and DSP for TALENT, Phoenix Mercury, Phoenix Venus, Phoenix Venus V2, Phoenix Venus V3, Phoenix Venus V4 and Phoenix Venus V4+). For those equipments that are defined in the system but are not connected the firmware versions are not shown. This information can be useful to AEQ Technical Service for maintenance purposes.



ControlPHOENIX

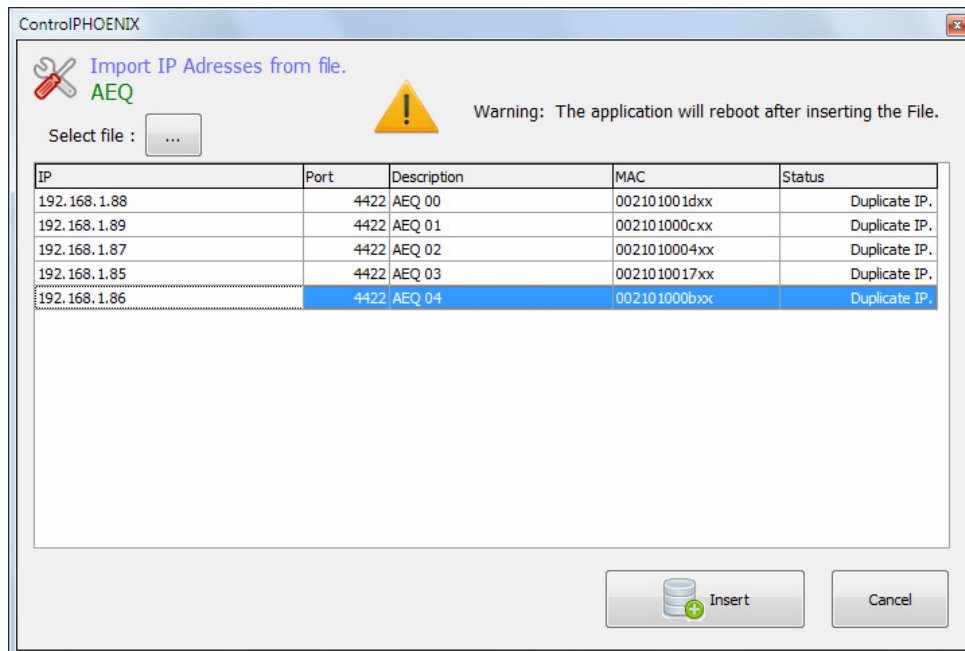
AEQ ControlPHOENIX: 2.2.2.9

AEQ

	Description	IP	CPU	FPGA	DSP
	AEQ 00	192.168.1.88	11.00 09/07/24	05.64 14/02/22	03.36 24/09/21
	AEQ 01	192.168.1.89	11.00 09/07/24	05.64 14/02/22	03.36 24/09/21
	AEQ 02	192.168.1.87	11.00 09/07/24	03.07 02/20/18	03.36 24/09/21
	AEQ 03	192.168.1.85	11.00 09/07/24		03.36 24/09/21
	AEQ 04	192.168.1.86	11.00 09/07/24		03.36 24/09/21

- "Load Equipment List from File".

This tool allows you to import a list of IP addresses from a CSV file in order to quickly incorporate new equipments to control. Once the file is loaded, the IPs list is shown on screen:



ControlPHOENIX

 Import IP Adresses from file.

AEQ

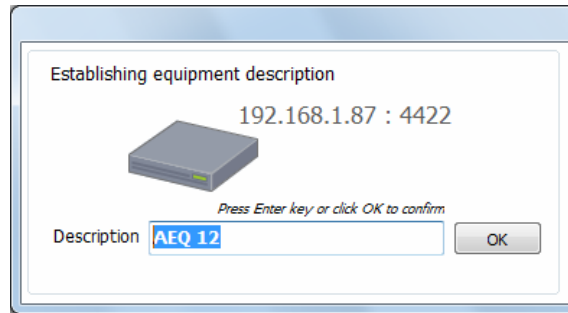
Select file :

 Warning: The application will reboot after inserting the File.

IP	Port	Description	MAC	Status
192.168.1.88	4422	AEQ 00	002101001dxx	Duplicate IP.
192.168.1.89	4422	AEQ 01	002101000cxx	Duplicate IP.
192.168.1.87	4422	AEQ 02	0021010004xx	Duplicate IP.
192.168.1.85	4422	AEQ 03	0021010017xx	Duplicate IP.
192.168.1.86	4422	AEQ 04	002101000bxx	Duplicate IP.

 Insert

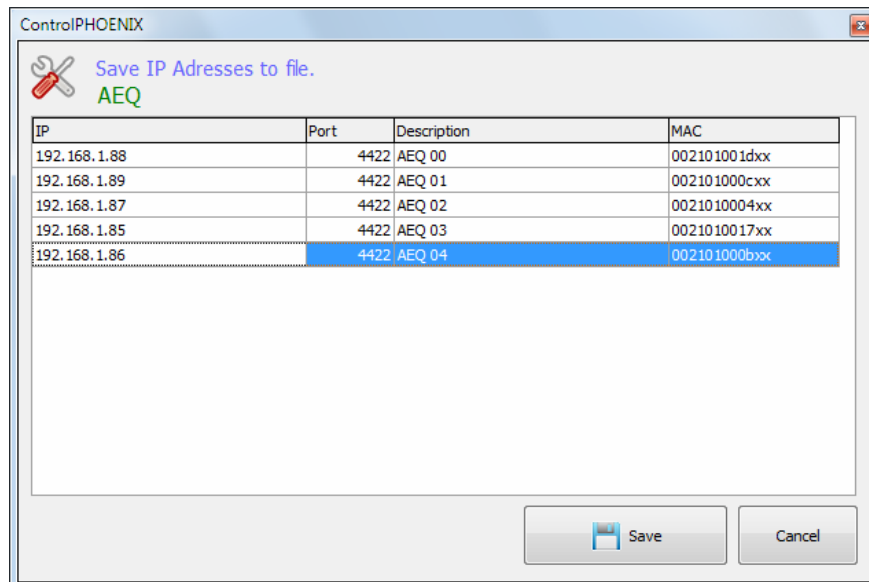
After confirmation by pressing the "Insert" button, several consecutive screens will appear (as many as IPs on the list) where you can confirm the name assigned to the unit by default or define a new one:



Once the equipments names are confirmed/defined, the application is closed. When you start it up again, autodiscovering is launched and, after selecting the ones you want to add, those units will appear available in the application plus the ones added by means of the text file.

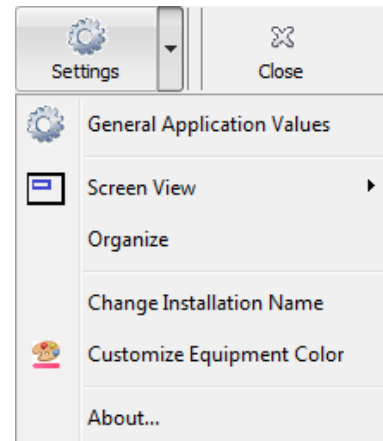
- **"Save Equipment List to File".**

This tool allows you to export the IP addresses of the different codecs (as well as the associated port, description and MAC) to a CSV file. This can be useful for network maintenance tasks.



5.1.10. Settings.

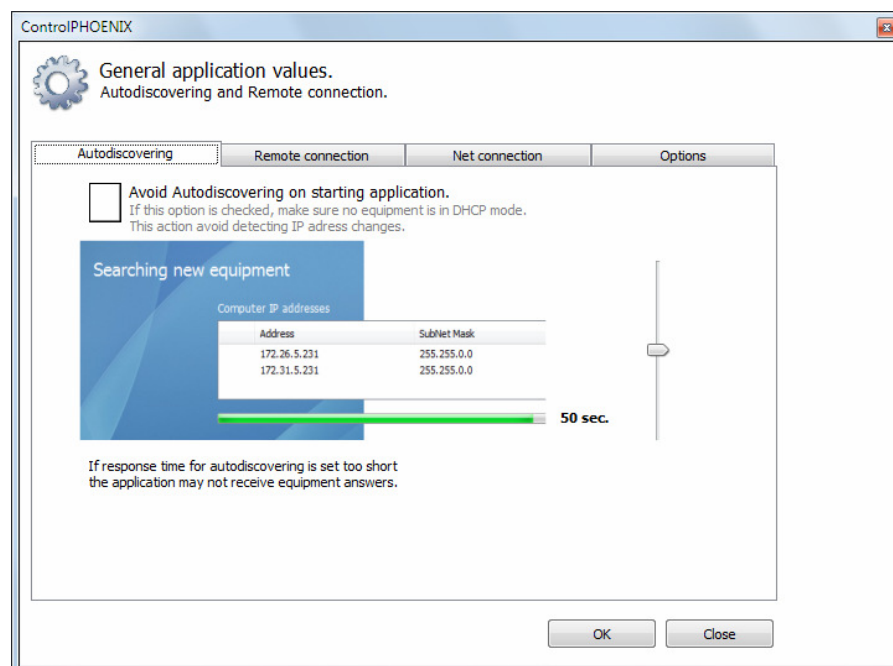
This menu gives access to several important configuration options:



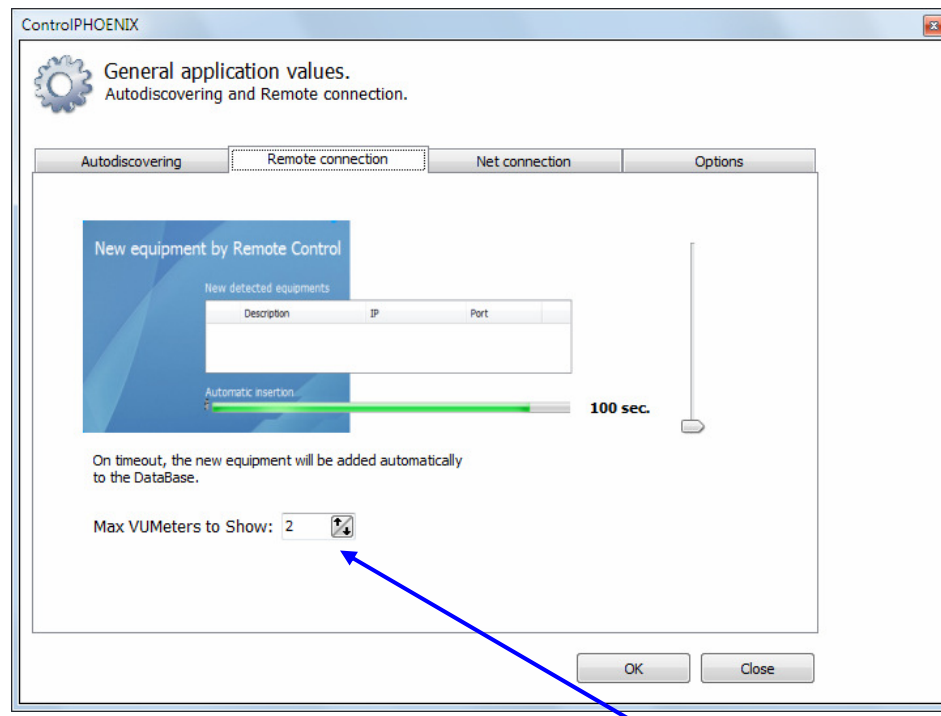
5.1.10.1. General Application Values.

This screen allows you to adjust the properties of the communications between the application and the equipment (see sections 4.1 and 4.2). There are 4 tabs: "**Autodiscovering**", "**Remote connection**", "**Net connection**" and "**Options**" that are described below:

- "**Autodiscovering**": This screen includes a check box that enables or disables whether the application will search for new equipment within the same network each time it is started. There is a control bar at the right that allows manual adjustment of the communications timeout, as a function of the networks' capacity.

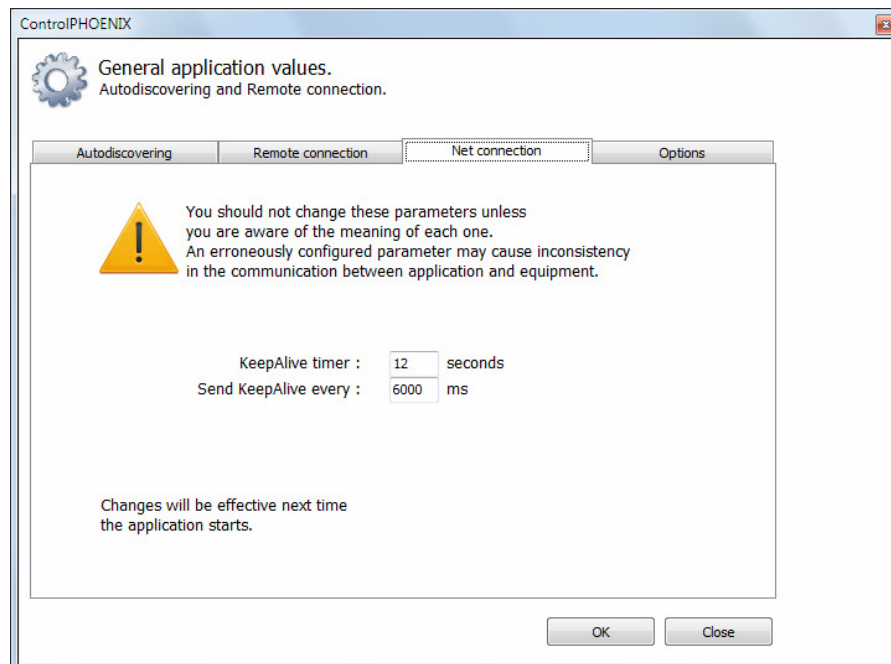


- "**Remote connection**": This tab is used to select how long the application delays the request from a piece of equipment to be included in the system before being accepted automatically if nobody cancels that.

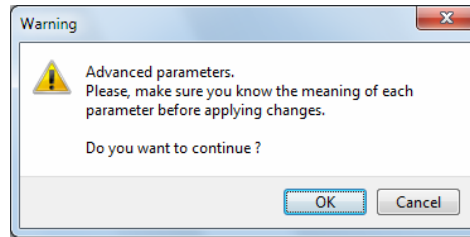


This tab also allows you to **modify the maximum number of vumeters** that can be simultaneously displayed. Application must be closed/open to make change effective.

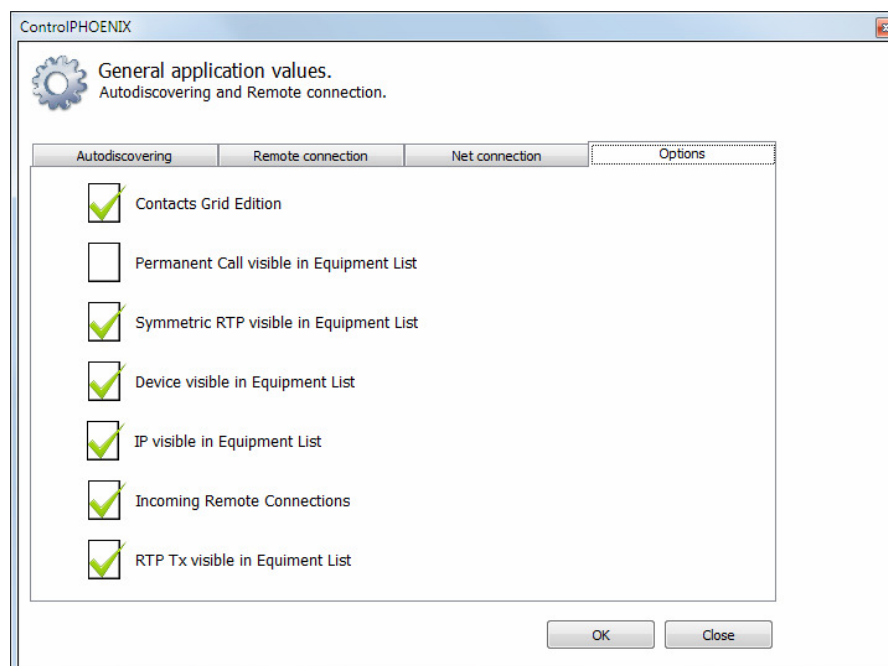
- **"Net connection"**: Two parameters are adjusted:
 - **"KeepAlive timer"**: timeout to disconnect the application and the unit if no incoming traffic is received
 - **"Send KeepAlive every"**: time the application will send traffic to the unit in order to avoid its disconnection.



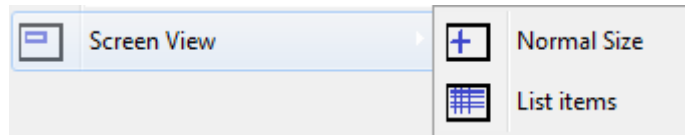
Due to the importance of both parameters, before configuring them a confirmation window will appear:



- **"Options"**: This screen includes the following check boxes:
 - **"Contacts Grid Edition"**: when activated allows you to edit a contact directly in the list of the contacts main screen.
 - **"Permanent Call visible in Equipment List"**: when activated a check box is shown in the "List items" view that allows you to activate/deactivate the automatic redial in any of the channels of a unit.
 - **"Symmetric RTP visible in Equipment List"**: when activated a check box is shown in the "List items" view that allows you to activate/deactivate the "Symmetric RTP" option in any of the channels of a unit.
 - **"Device visible in Equipment List"**: when activated the "Device" column is shown in the "List items" view: it indicates the device type.
 - **"IP visible in Equipment List"**: when activated the "IP" column is shown in the "List items" view: it indicates the device IP address.
 - **"Incoming Remote Connections"**: when activated allows units to connect remotely to the application (see section 4.2 of this manual).
 - **"RTP Tx visible in Equipment List"**: when this option is active, a new column is added in the list display mode for each channel indicating whether RTP transmission is active.

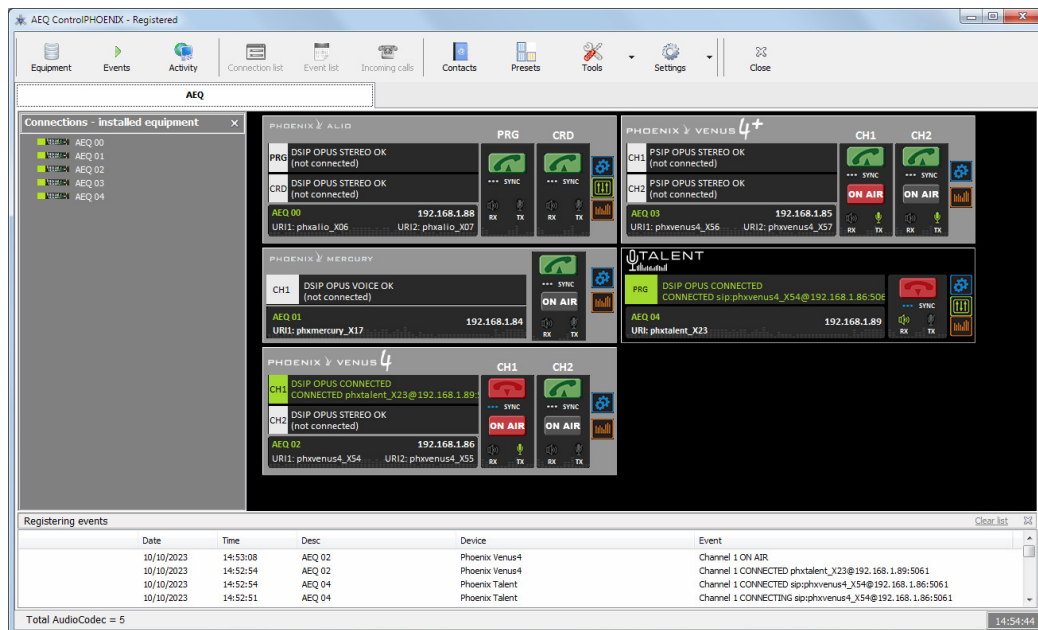


5.1.10.2. Screen View.

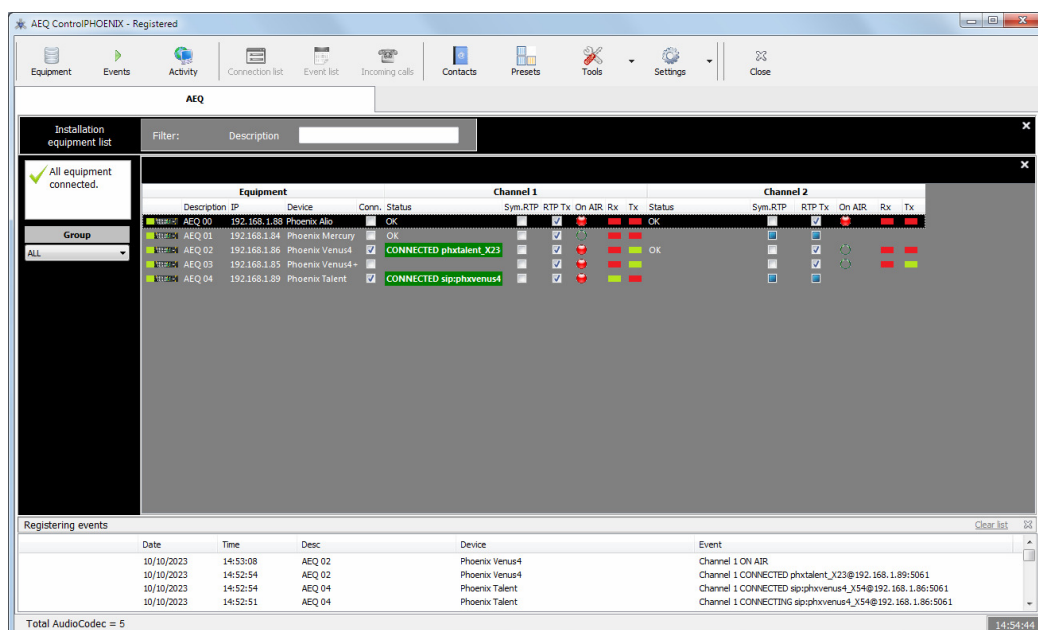


There are two ways to represent each individual codec in the screen:

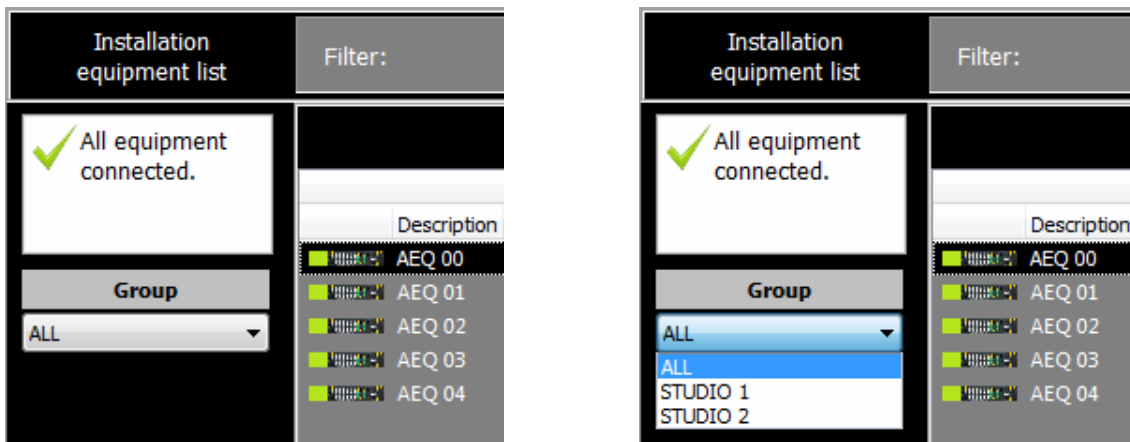
- **"Normal size"**: this mode opens an individual control window for each codec in the system.



- **"List items"**: this mode presents the units in a list. It's ideal for installations with many controlled equipment. By double-clicking on the line corresponding to one of them, the control screen for it will be displayed (this screen will be replaced by the corresponding to the next equipment the user double-clicks on in the list). The "Filter: Description" box allows you to filter the list by the characters that appears in "Description" column. Some of the columns and check boxes that are shown can be configured: see "Options" in section 5.1.10.1 of this manual.



In the case of the **"List items"** option, when **groups of units** have been previously created, it's possible to select which group you want to visualize by means of the drop-down menu at the left side. For more information, please consult section **"5.1.1. Equipment"** of this manual.

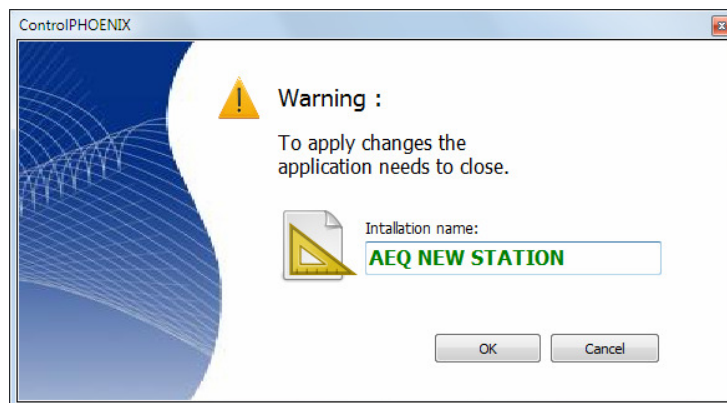


5.1.10.3. Organize.

This option arranges each codec control window within the screen in a tidy manner when "Normal Size" option is selected on "Screen View".

5.1.10.4. Change Installation Name.

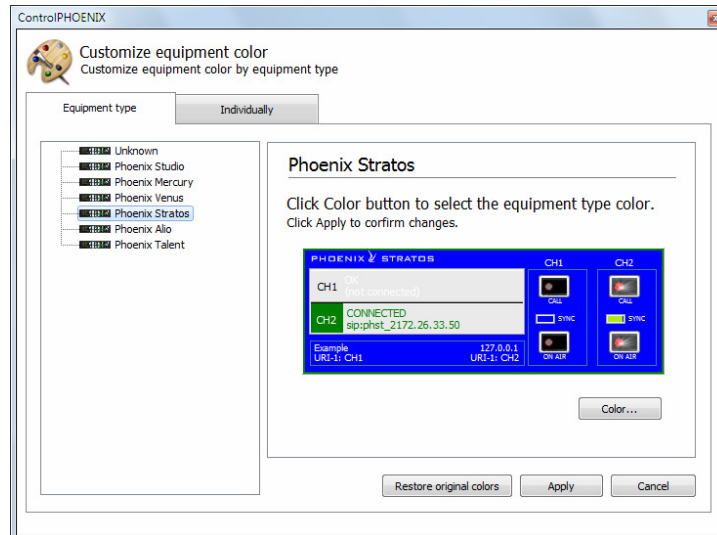
This option allows you to change the installation name. From then on, all equipment incorporated to control when the application starts will have this name followed by consecutive digits. Of course, these names can be edited at any time.



5.1.10.5. Customize Equipment Color.

This option allows you to change the background color of the registered units, so they can be distinguished at a glance by type, zone, priorities, etc. That configuration can be done by "Equipment type" or "Individually". The assigned color, such as the name, is saved on the unit itself, so that it will appear with the same color if it is accessed from another PC.

After clicking on "Equipment type", you must select a device type on the left list and the desired color (by pressing the "Color..." button), press the "Apply" button and confirm. Then the selected color will be applied to all the equipments of the same type (Phoenix Studio, Phoenix Mercury, Phoenix Venus, Phoenix Stratos, Phoenix Alio or Talent). There is a special type "Unknown" that is automatically created by the application if a new equipment was inserted but connection to it was never actually established, so its type cannot be determined until it finally connects.



When "Individually" tab is selected, you must select the desired equipment on the left list and the desired color (the same way as described before), press the "Apply" button and confirm. Then the selected color will be applied to the selected equipment.

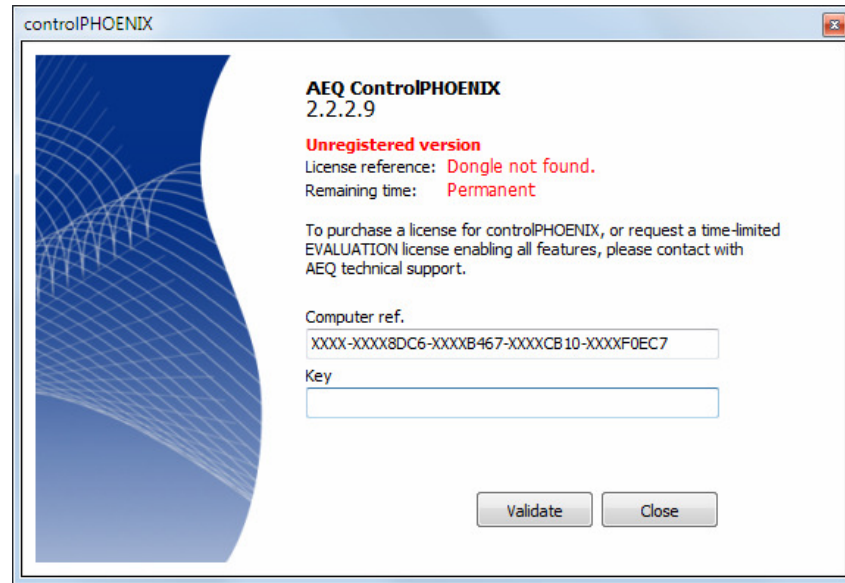
In both cases, the "Restore original colors" button allows you to return to the original colors of the units. Confirmation is also requested.

5.1.10.6. About.

This option shows information about the application version, as well as the AEQ contact data.

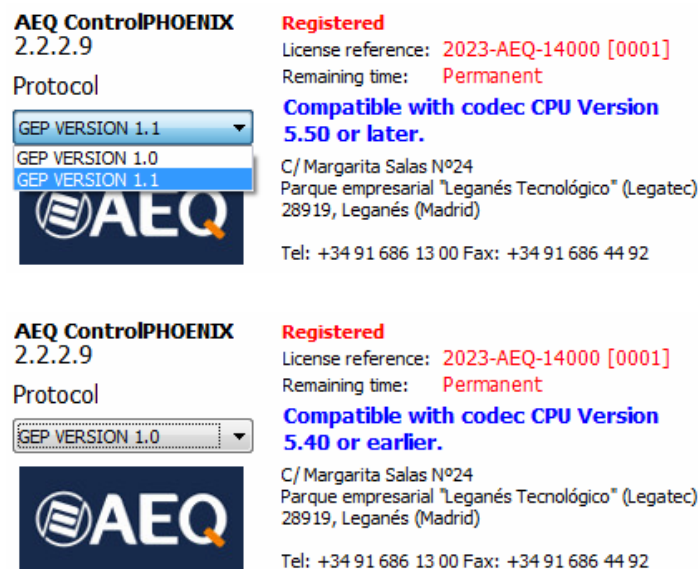


This option also indicates whether the application is **registered** or not, that is, if the **license** that allows you to control **more than 2 audiocodes** at the same time is purchased or not. The "**License**" button gives you access to the window where this option can be activated: once the license is purchased, after providing AEQ with the "Computer ref." that appears on that screen, the "Key" will be supplied in order to activate that functionality (in that case the license is associated with the computer where the application is installed). In order to activate that licence as well as for controlling several audiocodes from the application, it must be done from a PC user with **administrator** rights. In addition, in order to license the PC, it must have an assigned **MAC address**, which implies that it must be connected to a network. It's also possible to activate the license by means of a **Hardlock key**.



This option also indicates whether the application uses **GEP 1.1** protocol (it's the protocol configured **by default** when application is installed and allows you to control audiocoders with CPU version **5.50** or higher) or **GEP 1.0** protocol (allows you to control audiocoders with CPU version lower than **5.50**).

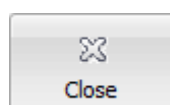
This window also allows you to modify the used protocol: application must be closed/open to make change effective.



It's NOT possible to control units with version 5.50 or higher and units with previous versions SIMULTANEOUSLY. In that case, these last ones should be upgraded to version 5.50 or higher.

If you have any doubt, please consult the AEQ Technical Service (support@aeq.es) or authorized distributors.

5.1.11. Close.



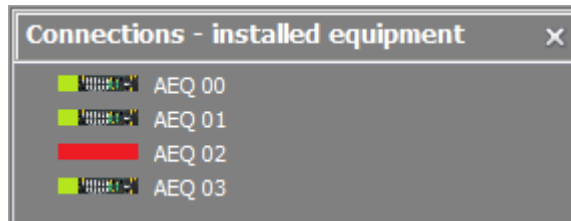
This button closes the application. Confirmation is requested.

5.2. Connection List.



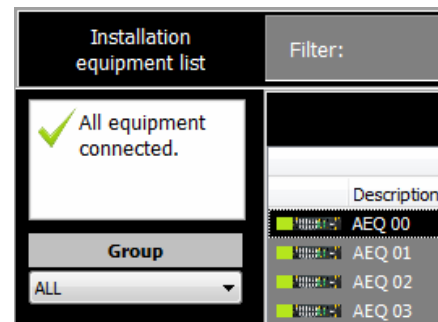
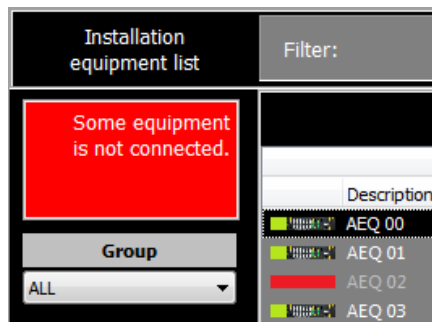
There is a window showing connections general status in the centre-left zone of the screen:

- In "Normal Size" mode, equipments that are not currently connected are marked in blinking red. The connected ones are marked in green.

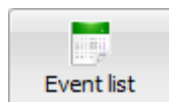


The connections list can be hidden by clicking  in the top right corner. In order to show it again, just click on the "Connection list" icon at the upper menu.

- In "List items" mode, a red blinking indicator on the left warns you whenever some equipment is not connected. When all defined units are connected this indicator is green.



5.3. Event list.



At the bottom part of the main screen, an event log is shown. The events can be cleared by using the "Clear list" button (confirmation is requested). The list can be hidden by clicking  in the top right corner. In order to show it again, just click on the "Event list" icon at the upper menu.

Registering events						Clear list	
	Date	Time	Desc	Device	Event		
	23/11/2018	8:54:07	AEQ 03	Phoenix Venus4	Channel 1 changed		
✓	23/11/2018	8:54:07	AEQ 03	Phoenix Venus4	Channel 1 ON AIR		
	23/11/2018	8:54:04	AEQ 03	Phoenix Venus4	Connected		
✗	23/11/2018	8:53:38	AEQ 03	Phoenix Venus4	Disconnected		
	23/11/2018	8:53:38	AEQ 03	Phoenix Venus4	No KeepAlive		
	23/11/2018	8:52:27	AEQ 03	Phoenix Venus4	Channel 1 OK		
	23/11/2018	8:52:27	AEQ 02	Phoenix Stratos	Channel 1 OK		
	23/11/2018	8:52:24	AEQ 03	Phoenix Venus4	Channel 1 DISCONNECTING sip:phxstratos_X26@192.168.1.87		
Total AudioCodec = 4						08:56:01	

5.4. Lower information bar.

Right below the events list, this bar indicates the number of registered codecs that are being controlled and the system time. In addition, an indicator shows any activity in the system.

Total AudioCodec = 4

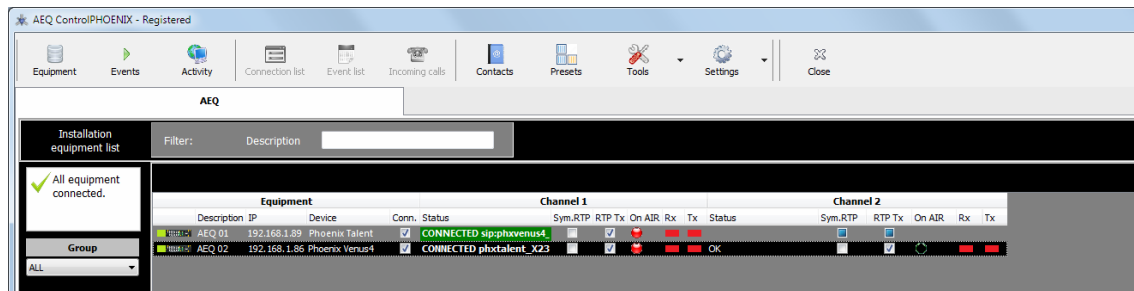
Activity 09:43:50

5.5. Connection statistics.

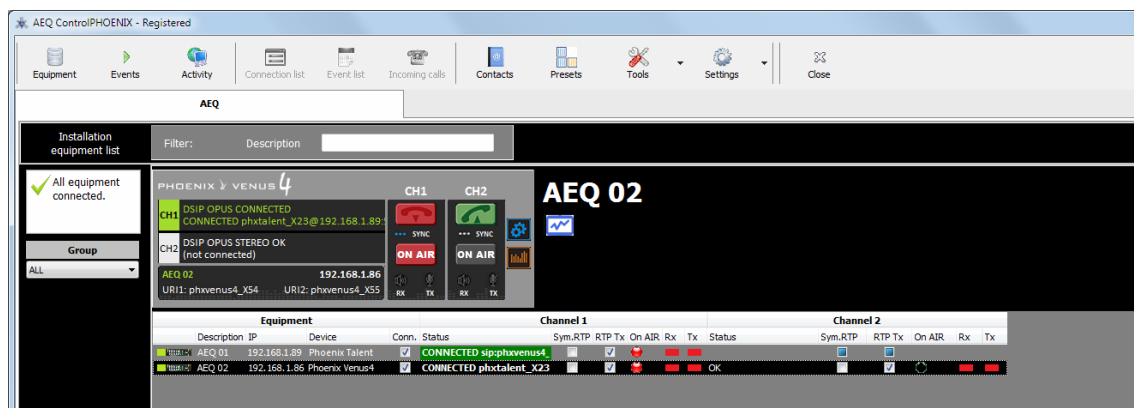
Version **2.2.1.54** (or higher) of **AEQ ControlPHOENIX** application, when working with audiocoders with CPU version **9.60** or higher, allows access to reception statistics of audio over IP connections (RTP or SIP). Those statistics consist of an elaboration and appropriate presentation of the data that was already available in textual format in previous firmware versions within the "STATUS" section in the audiocoder web server.

Access to statistics.

To access them, it is necessary to change representation to List mode ("Settings" → "Screen View" → "List items").



Once the unit for which we want to have statistics of its reception has been chosen, double-click on it to display its expanded view:



An icon with a graphic appears at the right of the unit, under its name:



By clicking on this icon, a couple of graphs and some counters will appear at the right of the unit:



Description.

The upper graph shows the size in milliseconds of the reception **buffer** of both channels (if the device has two, **CH1** in **red** and **CH2** in **blue**), along a sliding window 3 minutes long.

The lower graph shows the **jitter** value (variation of arrival delay of received packets), in milliseconds, synchronously to the upper window, also for both channels and with the same colors.

On the right, the values under the red label (**CH1**) show the type of buffer ("Buffer": FIXED or ADAPTIVE) and its configuration (size in the case of fixed buffer, maximum/minimum in the case of adaptive buffer). Below is the current size of that buffer, the accumulated number of lost packets ("Lost packets") and the total duration of the call ("Call duration ") in hh: mm: ss (24h format). The values under the blue label (**CH2**) represent the same, for the second channel.

By clicking on the **CH1** or **CH2** label it is possible to directly access the buffer configuration window for that channel.

If a channel is not connected, all its values and graphs will appear at 0. When starting a call, the duration and lost packets counters will also be reset to 0.

In the upper window, the packet loss events are shown by vertical cursors (boxed in red in the example). Those for CH1 are shown with an orange cursor, while those for CH2 are shown with a purple cursor, just like their respective lost packets counters.

Scales.

The scale of both graphs can be changed by clicking on the scale that appears at the left of the graph. The graph of the last 3 minutes will be redrawn according to the new scale. The lower value will always be 0.



If the buffer size plot does not appear when you open the statistics, for example, it is possible that the scale is too small, so you'll need to click on the scale at the left of the graph and enter a larger number (1500, for example). This way the scale will go from 0 to 1500ms. The same can be done with the jitter graph, and in the same way the scales can be reduced in order to observe small values in more detail.

Every time we open a statistic, by default the buffer scale will be from 0 to 200ms, and the jitter scale from 0 to 40ms. This initial setting can be modified in the "ControlPHOENIX.ini" file (located in the same folder where the application is installed) by changing the following fields:

```
[STATS]
MAXBUFFER=200
MAXJITTER=40
```

The first field indicates the default upper scale of the buffer graph and the second that of the jitter graph. NOTE: Changes in the "ControlPHOENIX.ini" file only take effect after restarting the application.

Use of statistics.

By means of these graphical statistics it is easy to correlate the events of excessive jitter with the loss of received packets if the buffer is not large enough at that time. If the type of buffer is adaptive, we can see how it varies between its specified values, tending to be reduced to a minimum unless there is a growth in network jitter that forces the codec to raise its buffer, up to the maximum value specified.

This way it's easier to configure the buffer (type and maximum and minimum sizes) in such a way that packet losses and delay are minimized for a given network.

NOTE: Receive buffer settings are made independently for each channel in "CONFIG" → "General" → "I/F Setup" (Fixed: Buffer length / Adaptive: buffer max and buffer min), or directly by clicking on the **CH1** or **CH2** label at the right of the graphs.

IMPORTANT: The statistics and buffer settings of a device refer only to the reception direction, so in bidirectional communications it is necessary to check the settings in the codecs of both ends, or at least in the one that presents quality problems of the received audio (microcuts, etc.).

It is normal to have sporadic packet loss in real networks, especially over the Internet. Its effect on the output audio will be more or less perceptible depending on the type of encoding chosen, the frequency with which these losses occur and how many packets are lost at once. Increasing the reception buffer size usually overcomes reasonable network imperfections, but it has a negative impact on delay, so adjustments must be made carefully.

The use of the adaptive buffer minimizes this delay, but it is more sensitive to sporadic losses depending on the behavior of the jitter. If having a fixed delay is allowable, it is recommended to use a fixed buffer of size 5-10 times larger than the maximum jitter value observed during a real stable connection.

Connection statistics log. Interpretation.

Sometimes it is necessary to carry out a long-term analysis of the connections quality, beyond the last 3 minutes that the graphs allow us to see. For that, additionally and only while the statistics window is open, a log is generated in the "**stats**" folder located where the program is installed (by default, in C:\Program files\AEQ\ControlPhoenix).

The content of this file (with a .txt extension, whose name is made up of the name of the codec and then the date) is separated into columns that show the different values captured for both channels.

The first row of the file contains texts indicating the content of the columns:

AEQ 02_09_06_22: Bloc de notas											
Archivo	Edición	Formato	Ver	Ayuda							
Current Time	buftype1	dur1_s	buf1_ms	jitt1_ms	lostpk1	buftype2	dur2_s	buf2_ms	jitt2_ms	lostpk2	
08:57:41	FIXED	55	500	0	0	NOT CONN	0	0	0	0	
08:57:42	FIXED	56	500	0	0	NOT CONN	0	0	0	0	
08:57:43	FIXED	57	500	0	0	NOT CONN	0	0	0	0	
08:57:44	FIXED	58	500	0	0	NOT CONN	0	0	0	0	
08:57:45	FIXED	59	500	0	0	NOT CONN	0	0	0	0	
08:57:46	FIXED	60	500	0	0	NOT CONN	0	0	0	0	
08:57:47	FIXED	61	500	1	0	FIXED	2	1000	0	0	
08:57:48	FIXED	62	500	4	0	FIXED	3	1000	4	0	
08:57:49	FIXED	63	500	17	0	FIXED	4	1000	17	0	
08:57:50	FIXED	64	500	4	0	FIXED	5	1000	4	0	
08:57:51	FIXED	65	500	2	0	FIXED	6	1000	2	0	
08:57:52	FIXED	66	500	0	0	FIXED	7	1000	0	0	
08:57:53	FIXED	67	500	0	0	FIXED	8	1000	0	0	
08:57:54	FIXED	68	500	0	0	FIXED	9	1000	0	0	
08:57:55	FIXED	69	500	2	0	FIXED	10	1000	0	0	
08:57:56	FIXED	70	500	0	0	FIXED	11	1000	0	0	
08:57:57	FIXED	71	500	3	0	FIXED	12	1000	1	0	
08:57:58	FIXED	72	500	2	0	FIXED	13	1000	0	0	
08:57:59	FIXED	73	500	0	0	FIXED	14	1000	7	0	
08:58:00	FIXED	74	500	10	0	FIXED	15	1000	10	0	
08:58:01	FIXED	75	500	7	0	FIXED	16	1000	0	0	
08:58:02	FIXED	76	500	0	0	FIXED	17	1000	0	0	
08:58:03	FIXED	77	500	0	0	FIXED	18	1000	0	0	
08:58:04	FIXED	78	500	497	3	FIXED	19	1000	497	0	
08:58:05	FIXED	79	500	478	5	FIXED	20	1000	492	0	
08:58:06	FIXED	80	500	0	5	FIXED	21	1000	0	0	
08:58:07	FIXED	81	500	0	5	FIXED	22	1000	6	0	
08:58:08	FIXED	82	500	1	5	FIXED	23	1000	5	0	
08:58:09	FIXED	83	500	23	5	FIXED	24	1000	0	0	
08:58:10	FIXED	84	500	0	5	FIXED	25	1000	0	0	

The data lines are added with a cadence of 1 second (see "Current Time" column, which corresponds to the actual capture time of that line).

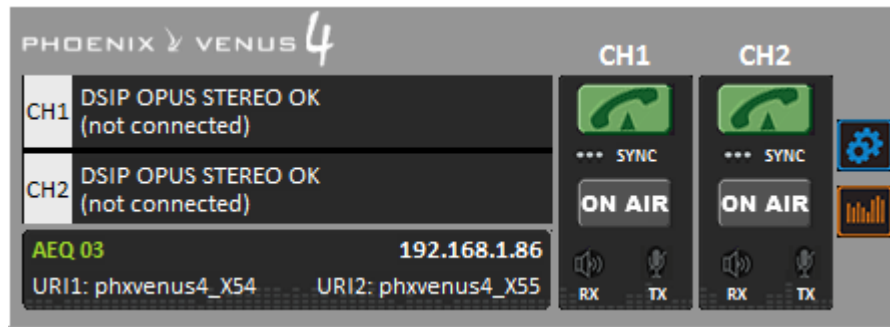
In the log of the previous example, we see that channel 1 is connected from the beginning of the capture (8:57:41). The type of buffer is "FIXED" all the time, the duration of the call ("dur1_s") increases, the buffer ("buf1_ms") remains at 500ms and the jitter ("jitt1_ms") varies. No packet loss ("lostpk1"=0) is detected until 08:58:04, where 3 packets are lost and then another 2, leaving "lostpkt1"=5 and being stable from that moment on. It can be seen that these losses coincide with a jitter increase to very large values, close to the buffer value (497 and 478ms).

Meanwhile, we can see that channel 2, to which the rightmost columns correspond, was disconnected at the beginning, so its buffer type was "NOT CONN" and the rest of values are 0, until 8:57:47, where it changes to "FIXED" and the counter starts to increment.

Data is easily importable into a spreadsheet for further analysis, graphic representation, etc. These files are generated by each unit and only while its statistics window is open.

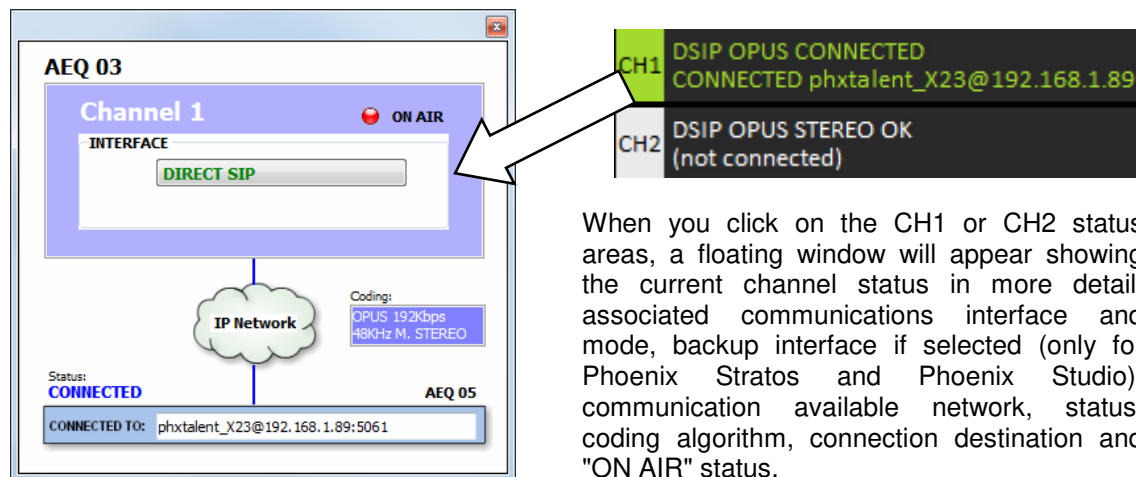
If the statistics window is closed, the file will continue to be filled only if it is reopened later for that same codec, but continuing with the new opening time, so a discontinuity will be observed in some fields such as the current time, duration of calls etc.

6. INDIVIDUAL CODEC CONTROL WINDOW.



This image represents the codec control window for a **Phoenix Venus V4**. The color of each codec window can be customized individually or by type as explained before in section 5.1.10.5.

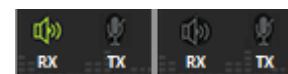
The central part of the screen represents the general status of each channel. At the right, a representation of the equipment "CALL" and "ON AIR" physical buttons and "SYNC" indicators for each channel of the current codec are displayed.



When you click on the CH1 or CH2 status areas, a floating window will appear showing the current channel status in more detail: associated communications interface and mode, backup interface if selected (only for Phoenix Stratos and Phoenix Studio), communication available network, status, coding algorithm, connection destination and "ON AIR" status.

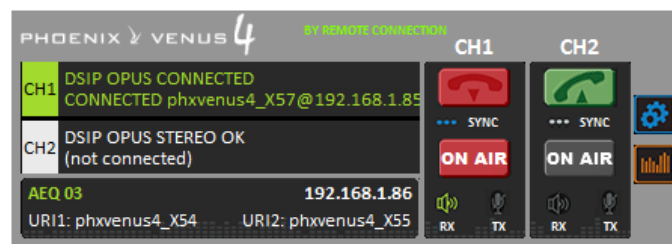
In this example, you can observe that channel 1 of equipment "AEQ 03" is associated to "DIRECT SIP", "ON AIR" is active and OPUS coding is being used. When the unit is connected to another codec that is also being controlled by the local application, its name is displayed ("AEQ 05" in this example). Otherwise, the name will appear as "Unknown".

There are also two audio input presence indicators per channel, for both transmission and reception.



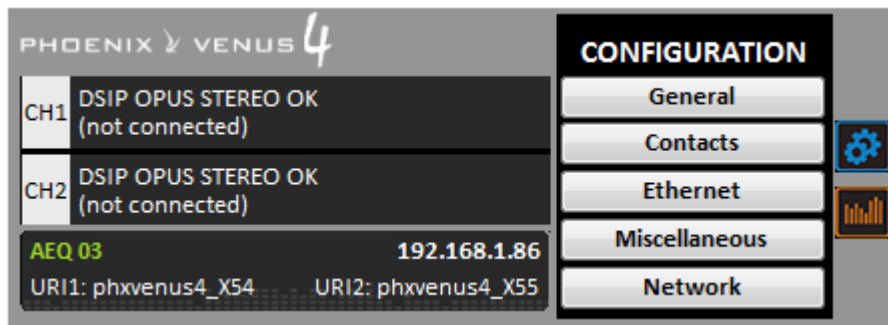
The "AUDIO" indicator will light green when audio is received. Thresholds and timeout can be individually adjusted in the main configuration screen.

Sometimes a piece of equipment has been incorporated to the control application by proper request, apart from the autodiscovery procedure described in section 4. In this case the individual codec control window will show the green label **"BY REMOTE CONNECTION"** beside the equipment type (see sections 6.5.3 and 7 of this manual).



Below the virtual screen the equipment is identified by its name, IP address and URI (per each channel). By clicking on the IP address an Internet browser will pop up showing unit's Web management window allowing, among other things, to update firmware and obtain real-time IP traffic statistics when the channel/s is/are connected.

At the right side, the  button gives access to a drop-down menu with the following options: "General", "Contacts", "Ethernet", "Miscellaneous" and "Network".

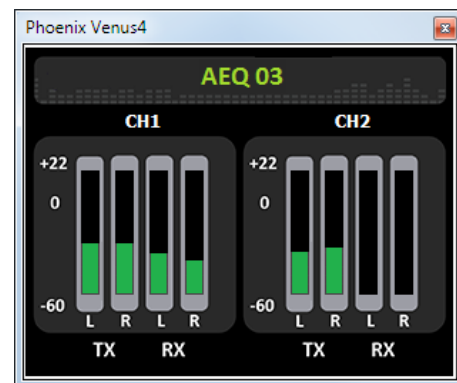


In order to close this menu, just press again that button.

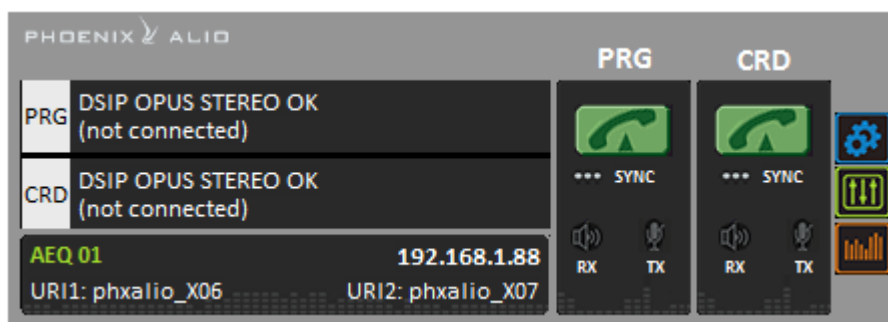
If you click on the  button, you will access to the equipments' real time vumeters, which will appear in a floating window that can be moved to the desired position. **By default**, it is possible to open up to **2 vumeters windows** in the application (for example, to check audio transmission between two connected equipments under the application's control). If you try to open a third vumeter window, the first one will be closed. Click on the right top cross in order to close a vumeters window.

The **maximum number of vumeters** that can be simultaneously displayed can be altered in the "Remote connection" tab of "General Application Values" (see section 5.1.10.1 of this manual).

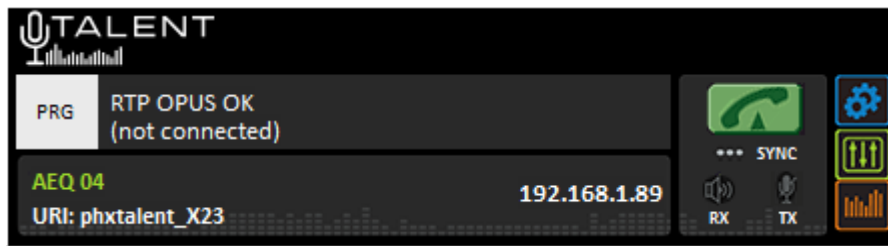
Vumeters represent audio transmitted to the L&R channels and audio received from the communication channel, also for both L&R. Except for Phoenix Mercury, the 2 available channels of the unit are shown (CH1 and CH2).



In the case of **Phoenix ALIO**, the individual codec window is slightly different. Instead of CH1 and CH2, the 2 channels are named **PRG** and **CRD** (Program and Coordination; when the unit has no **license**, the second channels appears deactivated); on the other hand, there are no "ON AIR" buttons and at the right side, in addition to configuration and vumeters buttons, a third button appears, that allows you to configure all the parameters of the ALIO mixer. For more information, please consult the Phoenix ALIO user's manual.



In the case of **TALENT**, the individual codec window is also different. There is only a channel named **PROGRAM**; on the other hand, as in the case of Phoenix ALIO, there is no "ON AIR" button and, at the right side, there are 3 buttons. For more information, please consult the TALENT user's manual.

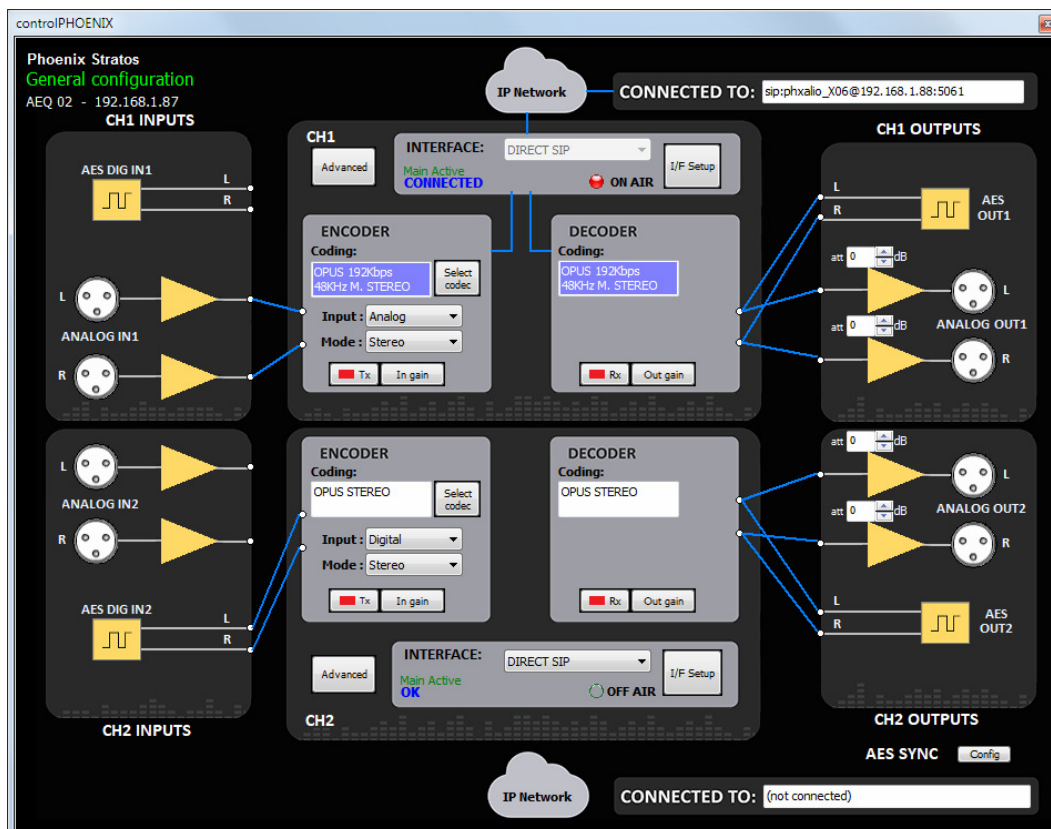


6.1. General configuration screen.

The general configuration screen provides access to the configuration of the main functions of the unit, such as connection of the audio inputs and outputs to the communication interfaces in order to get the most adequate audio transmission and reception.

From that screen you can configure the routing and audio levels in the unit, the encoding algorithms to use, the connection mode (in "INTERFACE" section) and you can access to the "Advanced" configuration and IP interface ("I/F Setup") menus.

The general configuration screen (for **Stratos** in the example) looks as follows:

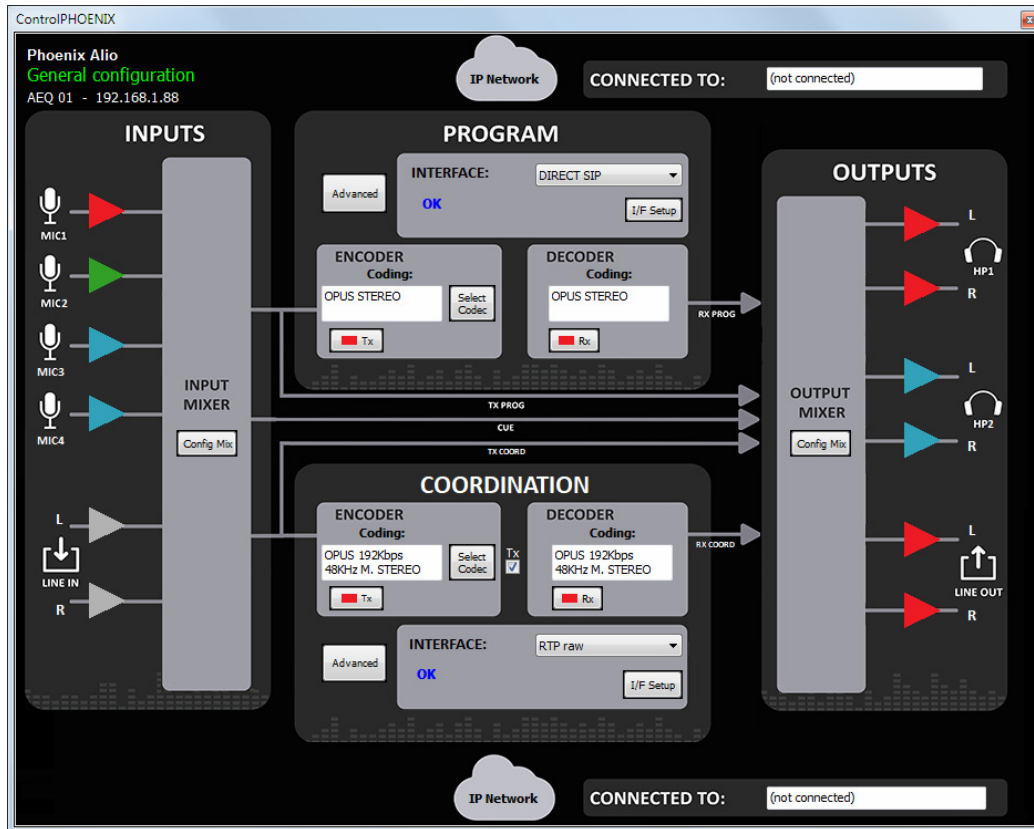


At the left, there is a block, "**INPUTS**", representing the analog and digital audio inputs of the equipment. In the central area you can find both channels ("**CH1**" and "**CH2**"), each one with its own ENCODER, DECODER and associated communications interface. On the top and on the bottom the communications networks are shown (IP, ISDN...) with the remote audiocoders. Finally, the "**OUTPUTS**" block at the right represents the equipment's analog and digital audio outputs.

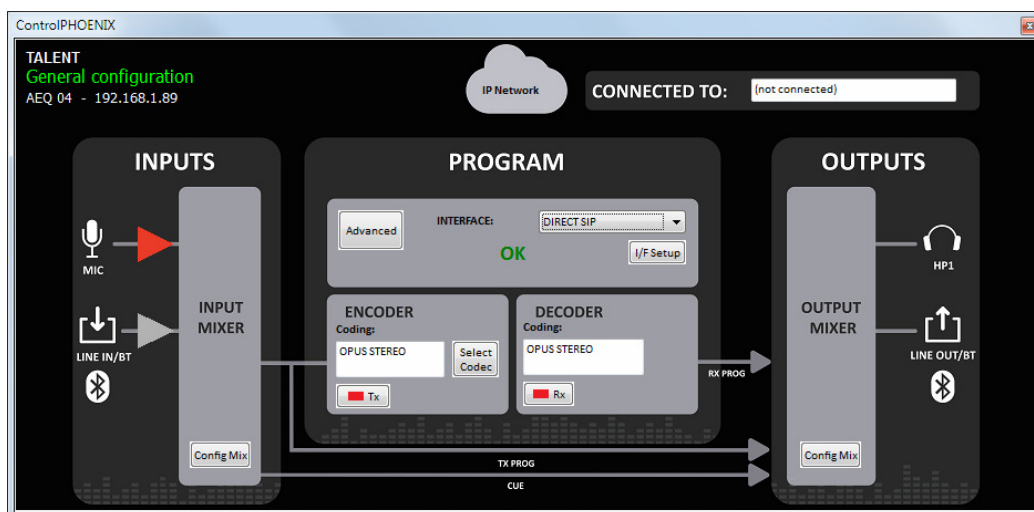
The diagram is updated automatically with each change in the input and output routing; the blue arrows will show the actual audio routing in every moment.

Each of the blocks appearing on the previous screen will be described in more detail in the following sections.

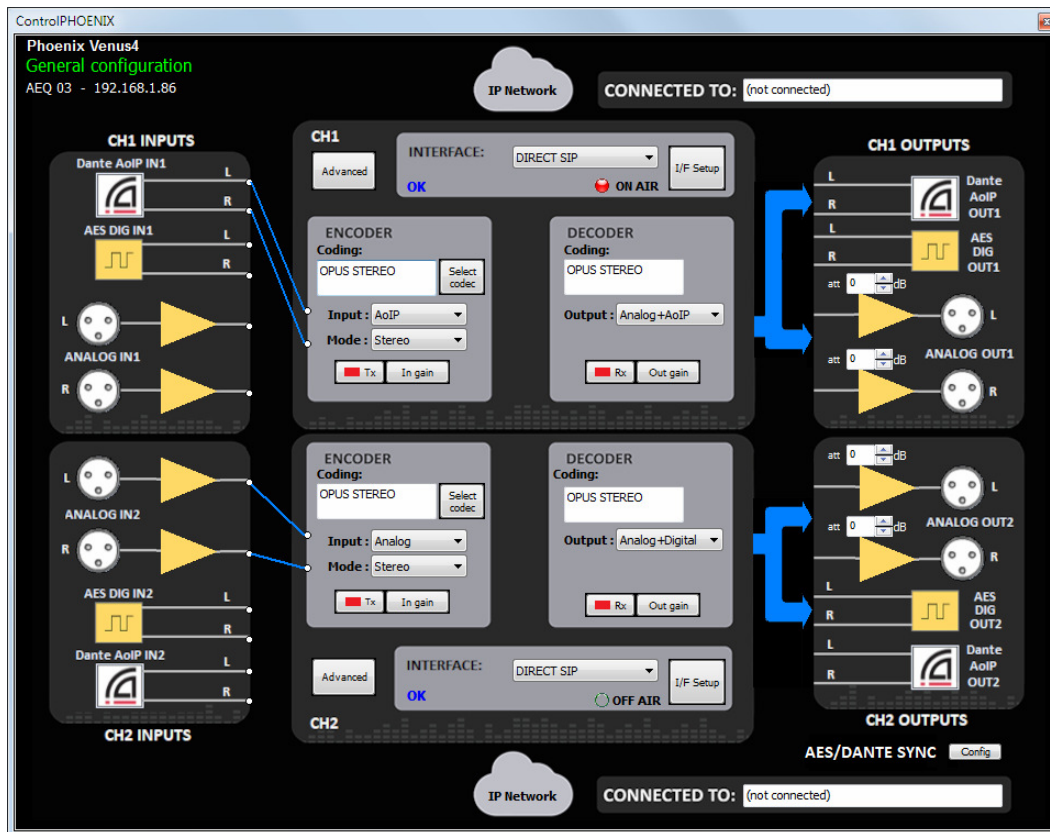
In the case of **Phoenix ALIO**, the general configuration screen is also slightly different. For more information, please consult the Phoenix ALIO user's manual (section 4.1.1).



In the case of **TALENT**, the general configuration screen is also different. For more information, please consult the TALENT user's manual (section 4.1.1).



In the case of **Phoenix VENUS V3, V4 and V4+**, the general configuration screen is also different, due to the possibility of activate the **DANTE AoIP** interface by means of a **license**. For more information, please consult the Phoenix VENUS user's manual (section 4.1).



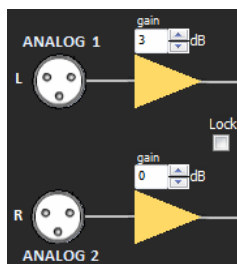
6.1.1. Inputs configuration.

The **"INPUTS"** block shows the equipment's audio inputs. There are two types of inputs (for Phoenix Mercury, Stratos, Studio and Venus):

- A stereo AES3 digital input per channel (only one channel for Phoenix Mercury).
- A stereo analog input per channel (only one channel for Phoenix Mercury). In the case of Phoenix Studio, there is only a stereo analog input per both channels, that can be assigned in stereo to one of the channels, can be distributed to both or used in independent mono mode (one input for each channel).



In the case of **Phoenix Venus V3, V4 and V4+**, a third type of optional inputs appears (it can be activated by means of a license): **AoIP DANTE**. For more information, please consult the Phoenix VENUS user's manual.



Only for Phoenix Studio, within this block you will find controls to adjust between 0 and +20 dB the **analog input gain**, independently for the right and left channels. It is possible to change the values for gain by clicking on the up and down arrows in the dial next to the number. If the **"Lock"** check box is ticked, both channels will receive the same value, i.e. are linked.

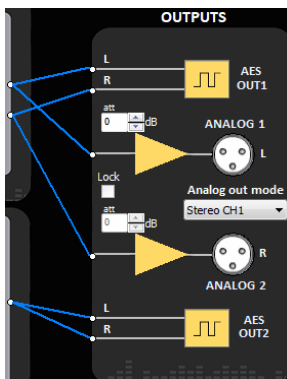


In the case of **Phoenix ALIO** there are four microphone inputs and one stereo line input. For more information, please consult the Phoenix ALIO user's manual.



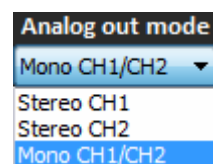
In the case of **TALENT** there is one microphone input and one stereo line input. There is also a Bluetooth input that is mixed with audio coming from the line input. For more information, please consult the TALENT user's manual.

6.1.2. Routing and attenuation configuration of the analog outputs.

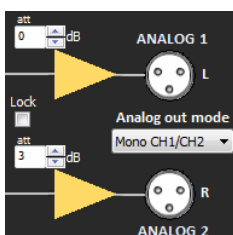


"**OUTPUTS**" block shows the equipment's audio outputs, both analog and digital. There is a stereo digital output and a stereo analog output per channel (only one channel for Phoenix Mercury).

In the case of **Phoenix Studio**, there is only a stereo analog output that can be associated in stereo to one of the channels, or be divided into two mono outputs (one for each channel). In order to configure this, there is a drop-down menu ("ANALOG OUTPUT MODE"):



In the case of **Phoenix Venus V3, V4 and V4+**, a third type of optional outputs appears (it can be activated by means of a license): **AoIP DANTE**. For more information, please consult the Phoenix VENUS user's manual.



This block also contains dials to adjust the **attenuation of the analog outputs** between 0 and 20dB, for both channels, independently. By clicking on the up and down arrows in the dial besides the number the values can be changed. In the case of **Phoenix Studio**, if the "**Lock**" check box is activated, both channels will receive the same value.



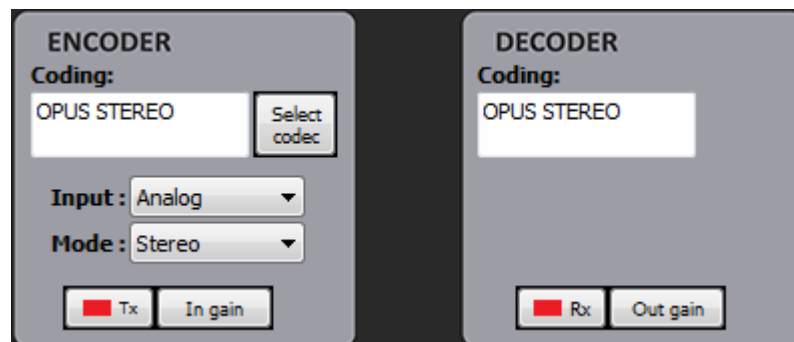
In the case of **Phoenix ALIO** there are two stereo headphones outputs and one stereo line output. For more information, please consult the Phoenix ALIO user's manual.

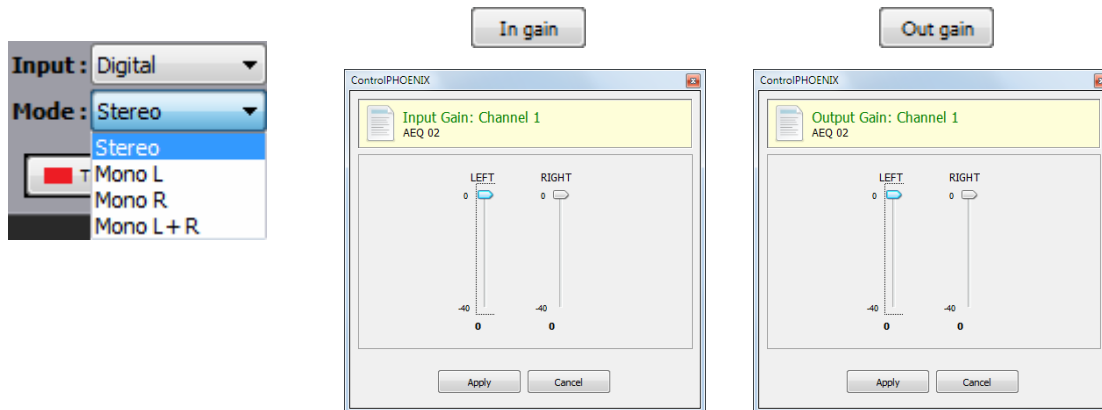


In the case of **TALENT** there is one stereo headphones output and one stereo line output. There is also a Bluetooth output. For more information, please consult the TALENT user's manual.

6.1.3. General audio configuration of a channel.

Each channel (encoder) receives audio (already converted to digital format) from the selected input. After this step, it can be further attenuated digitally by means of the "**In gain**" button. If a call is established, the audio will be encoded and sent to the communications interface. Conversely, audio received from the interface will be decoded. Then there is also an "**Out gain**" control that allows digital attenuation of the signal before being sent to the corresponding output.





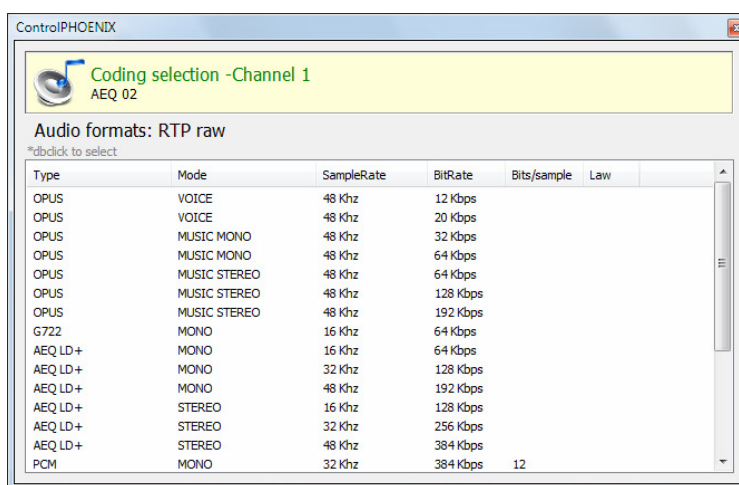
Therefore, each channel (CH1 and CH2) block provides controls to select the format of the audio source (analog or digital), select the audio input routing (stereo, mono L, mono R or mono L+R), adjust the digital attenuation level at inputs and outputs and select the coding/decoding formats. In the case of **Phoenix Venus V3, V4 and V4+** (only when Dante optional interface is activated by means of the corresponding license), "AoIP" input type can be also selected and you must select one of the 2 available output options: "Analog+Digital" or "Analog+AoIP" (please consult the Phoenix VENUS user's manual).

There is a field in the channel block showing the selected coding algorithm (or profile). When its background is white, this field is denoting the selected mode. When the background is blue, it is showing the mode actually used when a call has been established.



In the case of **Phoenix ALIO** and **TALENT**, the control of input and output levels is carried out from the mixer control window. For more information, please consult the Phoenix ALIO and TALENT user's manuals (section 4.1.2).

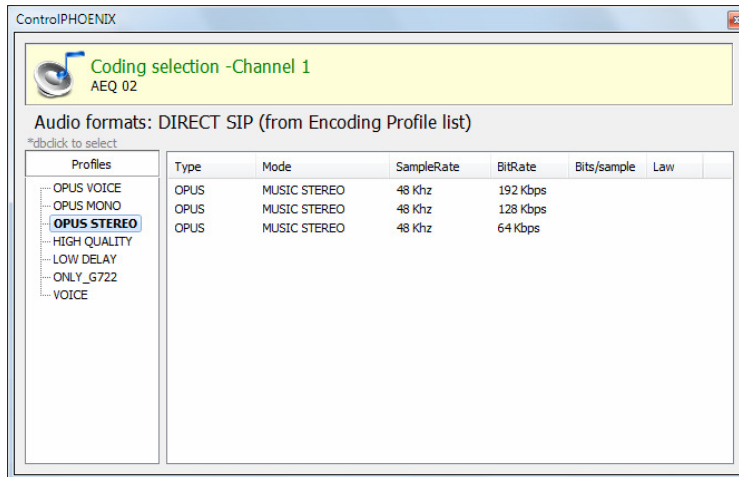
6.1.3.1. Coding selection.



By clicking on the **"Select codec"** button, the coding selection window will be displayed.

When "INTERFACE" is configured in RTP, ISDN or V35 mode, this list is already filtered in such way that only the modes that are compatible with the selected communications interface and configuration are visible.

The "DECODER" is automatically configured with the same coding mode.



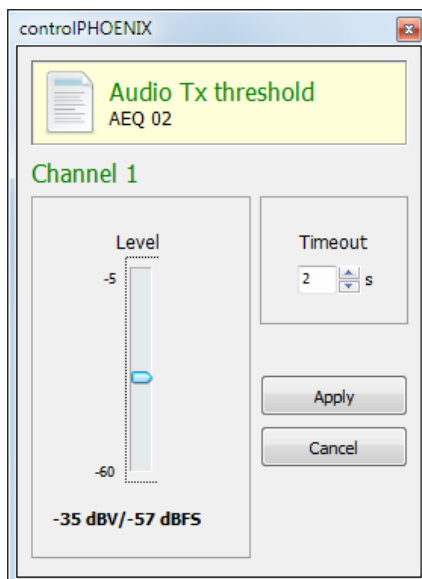
However, when the communications interface is configured in DIRECT SIP or Proxy SIP mode, encoding **"profiles"** will be displayed instead of particular coding modes. These profiles are lists of several coding modes. This is done this way because SIP allows the involved equipment to negotiate the coding to be used. The selection of codings during this process will be limited to the ones included in the selected profile only.

6.1.3.2. Audio detectors configuration.

Both the encoder and decoder blocks show a button ("Tx" and "Rx" respectively) with a visual indicator on it:



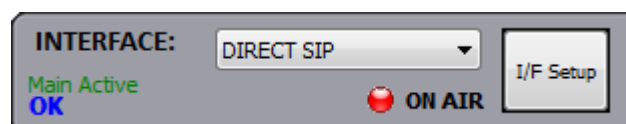
These indicators correspond to the audio input detector (in the case of the encoder block, "Tx"), that is, the audio to be sent, and the received audio detector (in the case of the decoder block, "Rx").



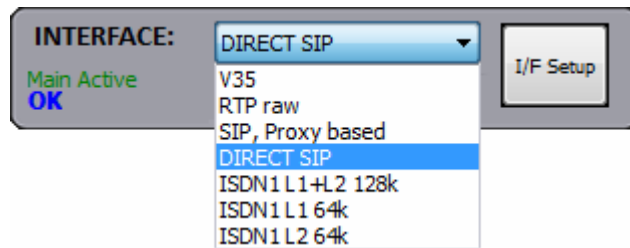
By clicking in these buttons, we can adjust both the audio thresholds (expressed in dBV for analog input and dBFS for digital input) and the timeout that the application allows to still announce audio presence even when audio has fallen below the threshold. Once this timeout has expired, the indicator will return to red.

6.1.3.3. Communications interface selection and configuration.

There is a section at each channel block that allows configuration of the communications interface the audio will be sent and received thru.

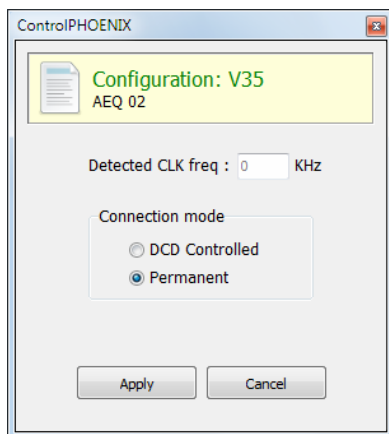


By clicking on the drop-down "INTERFACE" menu, the available interfaces will be displayed. The **V35** and **ISDN** options are only available for Phoenix Stratos and Phoenix Studio (it's optional for this one). The **IP** interface will appear in every possible operating mode: Proxy SIP based, SIP without Proxy (DIRECT SIP) and RTP raw (SIP protocol not used). By selecting a particular interface among all available ones and clicking on "**I/F Setup**" you will access to the configuration window. This window depends on the particular interface; all of them are described on next section.



6.1.4. Interfaces.

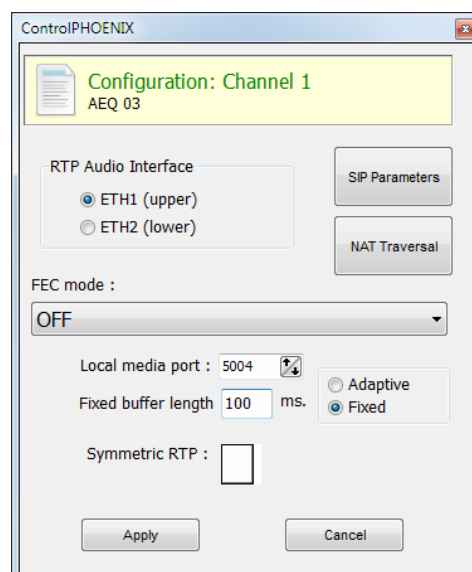
6.1.4.1. V35 communications interface configuration (Phoenix Stratos and Phoenix Studio only).



V35 interface configuration is limited to selection of the operating mode (permanent or DCD signal controlled) and an informative window that displays the frequency detected for the incoming clock.

6.1.4.2. IP communications interface configuration. SIP modes.

Channel 1 logical network interface is always denoted as "**Channel 1**" ("PRG" for Phoenix ALIO and TALENT). When there is a Channel 2 logical network interface (in the case of a Phoenix Stratos, Phoenix Studio, Phoenix Venus, Phoenix Venus V2, Phoenix Venus V3, Phoenix Venus V4 or Phoenix Venus V4+) is called "**Channel 2**" ("CRD" for Phoenix ALIO).



The configuration of the IP interface includes the following parameters:

- **"RTP Audio Interface"**: allows you to select the physical network interface (**ETH1** or **ETH2**) assigned to the channel audio. This option is only available for **Phoenix Studio**, **Phoenix Venus V2**, **Phoenix Venus V3**, **Phoenix Venus V4** and **Phoenix Venus V4+**.
- **"FEC mode"**: allows you to configure whether **FEC** (Forward Error Correction) is used or not and its type. Error correction is performed by sending redundant information that allows the receiver to recompose the lost data in case of not-perfect transmissions.

Forward error correction always generates a higher binary rate, and this in turn can generate more losses in very narrow transmission channels, as well as delays. It is recommended that the communication is started with no FEC (OFF) and, once established, experiment with the different available modes in case of problems and check if the results are better with some of them.

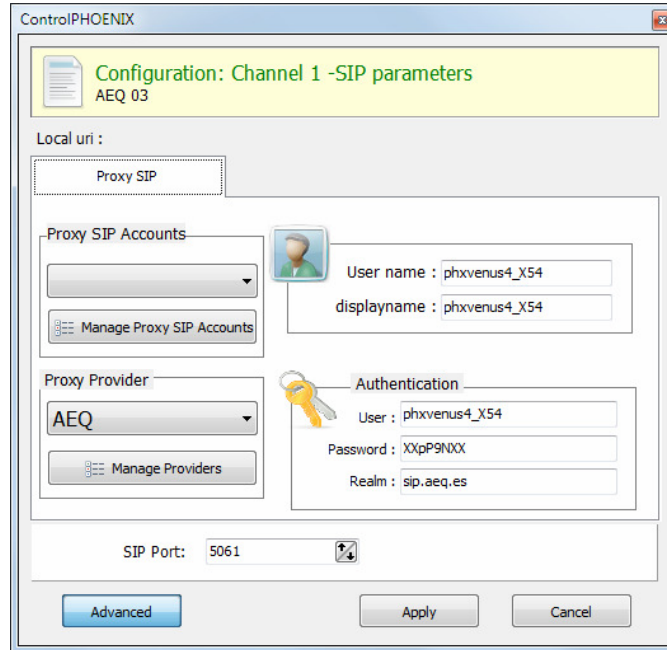
 - **LOWEST**: generates a 40% higher binary rate and produces a 575 ms additional delay.
 - **LOW**: generates a 50% higher binary rate and produces a 375 ms additional delay.
 - **MIDDLE**: generates a 66% higher binary rate and produces 225 ms additional delay.
 - **HIGH**: doubles the binary rate producing 125 ms additional delay.
- **"Local media port"**: allows you to configure the UDP port selected to transmit audio at origin over IP. Minimum value 1,024. Maximum value 65,534. Default value 5004.
- **"Adaptive/Fixed"**: allows you to configure the **reception buffer** as adaptive or fixed. In the first case, its size will vary according to the network transmission quality. In fixed mode, its size will be steady according to manual configuration.
- **"Adaptive buffer max"/"Fixed buffer length"**: this is the maximum size of the reception buffer. When it is defined as adaptive, Phoenix audiocodex will start to shorten it from this value as the network's transmission quality allows. If it is defined as FIXED, this max value will remain, as the buffer's size won't be varied during the connection. This value must be set in milliseconds. The longer the buffer is, packet misses will be less likely, but base delay will also be longer, specially if the buffer is set to FIXED mode. That maximum value is limited to 2000 ms.
- **"Adaptive buffer min"**: this is the minimum size of the reception buffer. This parameter only appears in the case of adaptive buffer.
- **"Symmetric RTP"**: this option allows you to force the local unit to send audio to the same IP and port from which it is receiving audio. The destination port specified when making the call will be ignored when we receive packets from the remote equipment. This option will allow you to connect to an equipment with unknown IP and/or port (because it's behind a router with NAT, for instance).

When in SIP mode ("SIP Proxy based" and "DIRECT SIP") we will additionally have access to the buttons to configure the parameters associated to SIP protocol ("SIP Parameters") and "NAT Traversal" strategies.

SIP parameters configuration (accessible by pressing **"SIP Parameters"** button) is slightly different depending on whether the channel is operating with Proxy ("SIP Proxy based") or without it (DIRECT SIP).

The common parameters to configure are the following ones:

- **"User Name"**: enables you to edit the name of the unit and how it will be reflected in the diverse internal menus of the unit.
- **"Display Name"**: editable name. This is the public name of the equipment that will be used in SIP server, so you can recognize the equipment with this identifier externally to the system.
- **"SIP Port"**: enables you to configure the SIP port. If this parameter is modified, it is recommended to restart the unit to ensure that the configuration takes effect, especially if there is another channel and it is also configured in some SIP mode.



Configuration: Channel 1 -SIP parameters
AEQ 03

Local uri :

Proxy SIP

Proxy SIP Accounts

Manage Proxy SIP Accounts

Proxy Provider

AEQ

Manage Providers

Authentication

User : phxvenus4_X54

displayname : phxvenus4_X54

User : phxvenus4_X54

Password : XXp9NXX

Realm : sip.aeq.es

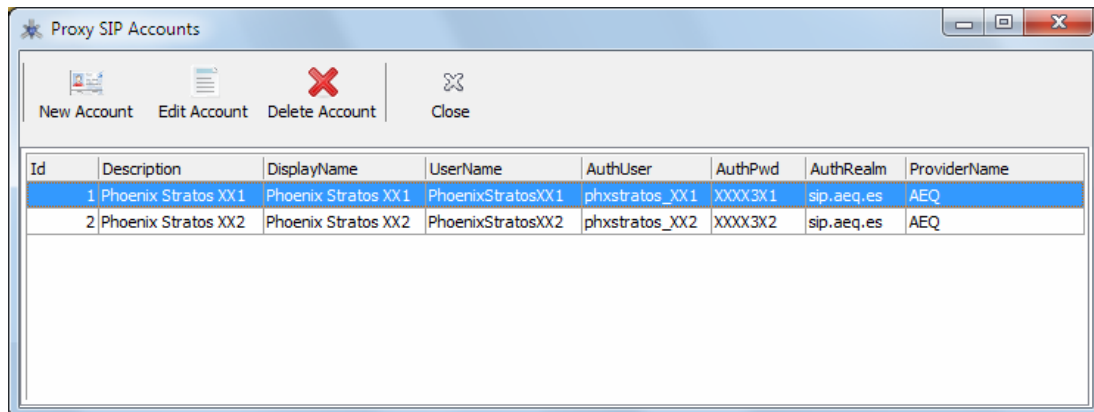
SIP Port : 5061

Advanced Apply Cancel

When using **"SIP Proxy based"** mode the following parameters are also displayed:

- **"Proxy SIP Account"**: enables you to select a Proxy SIP account from a previously created and stored list. In case an account is selected, the parameters described next would be automatically loaded (confirmation is requested). This parameter, just like "User Name" and "Display Name", only appears when pressing the **"Advanced"** button.
- **"Proxy Provider"**: enables you to select the external SIP server with which the unit will work from a previously stored list. By default, AEQ server will be selected.
- **"Authentication"**: enables you to edit the password and security information for the user profile associated with the unit in the previously selected SIP server. By default, the data configured in this field in order to use AEQ server are the following:
 - o **"User"**: the "User Name" configured in Factory, "phxvenus3_X56" for instance. In most cases, the name of the interface user and the authentication user will be the same (depending on the provider).
 - o **"Pwd"**: the Password associated to that user.
 - o **"Realm"**: the domain where the SIP Server is located, by default: sip.aeq.es.

The configuration of the Proxy SIP accounts list that appears available in "Proxy SIP Accounts" section is carried out in the window the **"Manage Proxy SIP Accounts"** button gives access to. See sections 5.1.7.1 or 5.1.8 of this manual for a detailed description.



Proxy SIP Accounts

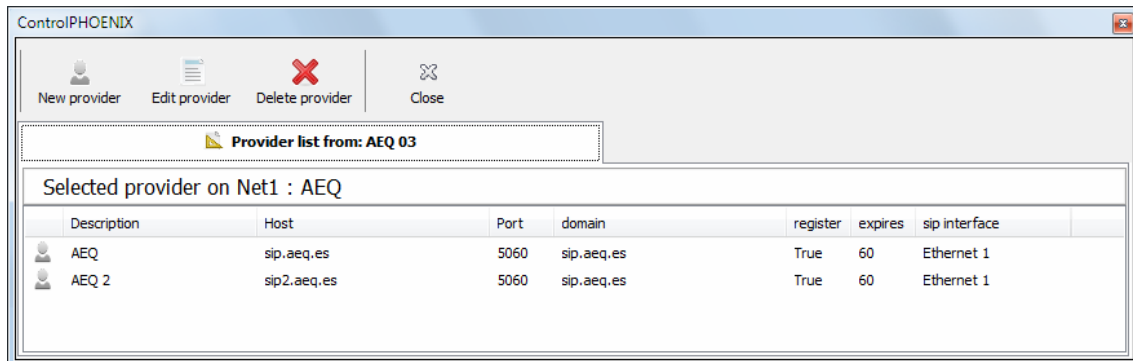
New Account Edit Account Delete Account Close

Id	Description	DisplayName	UserName	AuthUser	AuthPwd	AuthRealm	ProviderName
1	Phoenix Stratos XX1	Phoenix Stratos XX1	PhoenixStratosXX1	phxstratos_XX1	XXXX3X1	sip.aeq.es	AEQ
2	Phoenix Stratos XX2	Phoenix Stratos XX2	PhoenixStratosXX2	phxstratos_XX2	XXXX3X2	sip.aeq.es	AEQ

This window includes options to manage the different Proxy SIP accounts we want to have available (add, edit and delete). Data to be configured for each account is:

- Description.
- Display Name and User Name
- Authentication user and associated password.
- Domain where the SIP Server is located.
- Proxy Provider name.

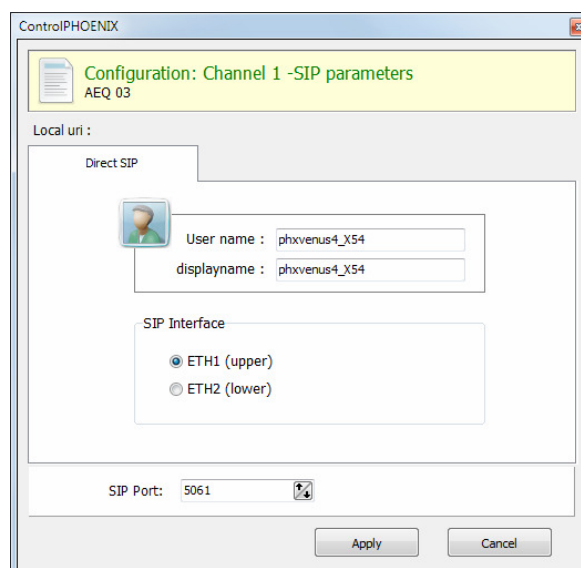
The configuration of the SIP servers list that appears available in "Proxy Provider" section is carried out in the window the **"Manage Providers"** button gives access to:



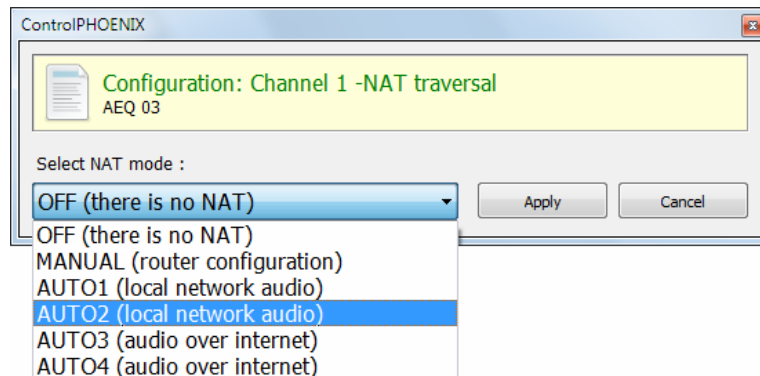
This window includes options to manage the different SIP providers we want to have available (add, edit and delete). Data to be configured for each provider is:

- Description.
- IP or name of the Proxy Server.
- Proxy SIP port (5060 by default).
- Domain the server attends (its own IP is also valid).
- Whether the server supports registration or not.
- Expiration time for the registrations, in seconds.
- Physical network interface (ETH1 or ETH2, just for Phoenix Studio) used for the communications with the proxy (it can be the same as the audio interface).

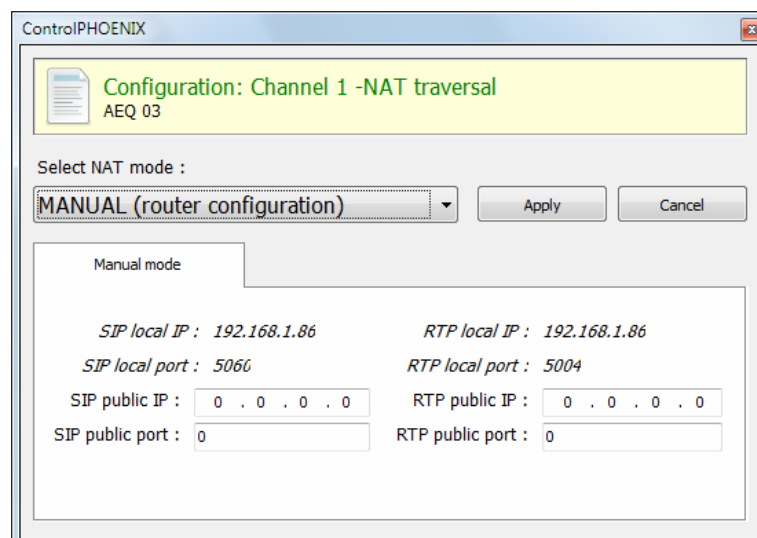
When using **"DIRECT SIP"** mode, in addition to "User Name", "Display Name" and "SIP Port" parameters, only network interface (ETH1 or ETH2) that will be used for SIP protocol must be configured in **"SIP Interface"** (this option only appears in the case of **Phoenix Studio**, **Phoenix Venus V2**, **Phoenix Venus V3**, **Phoenix Venus V4** and **Phoenix Venus V4+**).



Finally, we can configure the way the equipment will use to traverse NAT (Network Address Translation) routers by clicking on the **"NAT Traversal"** button. Six modes are available:



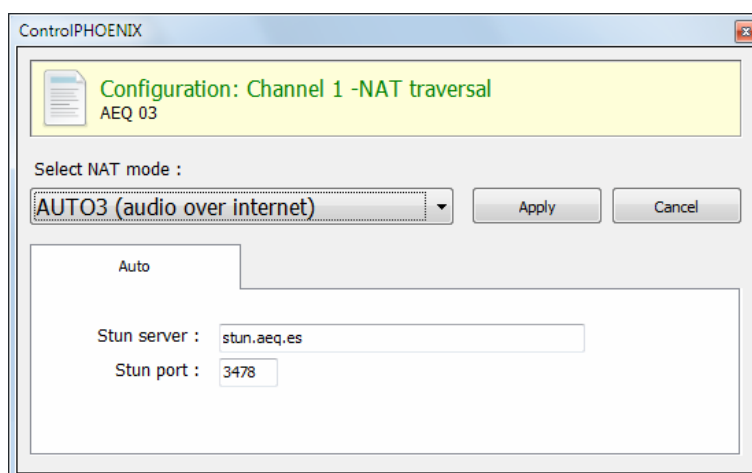
- **"OFF (there is no NAT)"**: this mode will not act with regards to traverse NATs. This mode will be used only to operate in the local network (all of the SIP participants are in the same local network, including the Proxy SIP, if we use it), but it is also useful when the unit operates only in the public Internet network.
- **"MANUAL (router configuration)"**: this mode (and next ones) are to be used when the unit is operating inside a private network but has to connect to other equipment in different networks through Internet. In this mode, the unit will not perform any action aimed to know how to access Internet in the best possible way. It will simply leave the user the responsibility to indicate the IP and public ports to be included in SIP messages. The user will have to know these parameters and may have to ask the Network administrator or IT manager for a port-forwarding in the firewall, directing to the IP and private ports of the equipment.



The eight parameters to be configured in the dialog for this mode are:

- **SIP LOCAL IP**: read-only parameter that tells you the IP of the IP interface of the unit as regards SIP, so that the latter can, in turn, convey this to the router or firewall administrator when it is configured. It can be set in order to adapt it to network necessities in menu: "Configuration" → "Ethernet".
- **SIP LOCAL PORT**: read-only parameter that tells you the port of the IP interface of the unit used for SIP signaling, so that the latter can, in turn, convey this to the router or firewall administrator when it is configured. Before checking the value of this parameter you should have configured previously whether you want to work with Proxy or not and restart the unit.

- **SIP PUBLIC IP:** parameter that will tell the unit which public IP will correspond to it, so that it can include the said IP in its SIP messages. The router or firewall administrator must tell you the value of this parameter.
 - **SIP PUBLIC PORT:** parameter that will tell Phoenix which public port it will have corresponding to its local SIP port. The router or firewall administrator must tell you the value of this parameter in order to make the required port-forwarding.
 - **RTP LOCAL IP:** read-only parameter that tells you the IP of the IP interface of the unit as regards RTP, so that it can, in turn, convey this to the router or firewall administrator when it is configured. You will usually configure the same network interface as for SIP, so it will be the one configured in point number 1.
 - **RTP LOCAL PORT:** read-only parameter that tells you the port of the IP interface of the unit as regards RTP, so that the latter can, in turn, convey this to the router or firewall administrator when it is configured.
 - **RTP PUBLIC IP:** parameter that will tell the unit which public IP will correspond to the RTP of its IP interface, so that it can send the said IP in its SIP messages. The router or firewall administrator must tell you the value of this parameter. Usually the administrator will take out SIP traffic and RTP using the same public IP configure in point number 3.
 - **RTP PUBLIC PORT:** parameter that will tell the Phoenix Studio which public port will correspond to the RTP of its IP interface, so that it can send the said port in its SIP messages. The router or firewall administrator must tell you the value of this parameter in order to make the required port-forwarding.
- **"AUTO 1 (local network audio)":** this mode is basically used when the equipment is going to register in a SIP proxy server through internet but it is going to connect to other audiocodecs inside the same local network. In this mode the SIP proxy is supposed to carry out the necessary task to traverse the NAT the equipment is behind of.
 - **"AUTO 2 (local network audio)":** this mode is similar to AUTO1 except that we no longer rely on the SIP proxy to carry out the NAT traversal tasks, the equipment will do that.
 - **"AUTO 3 (audio over internet)":** this mode is also equivalent to AUTO1 but the equipment will connect to other audiocodecs through Internet. This mode tries to deliver the audio through Internet with the least possible restrictions.
 - **"AUTO 4 (audio over internet)":** this model is equivalent to AUTO2 with the exception that the equipment will also connect to audiocodecs through internet also. This mode tries to deliver the audio through Internet with the least possible restrictions.

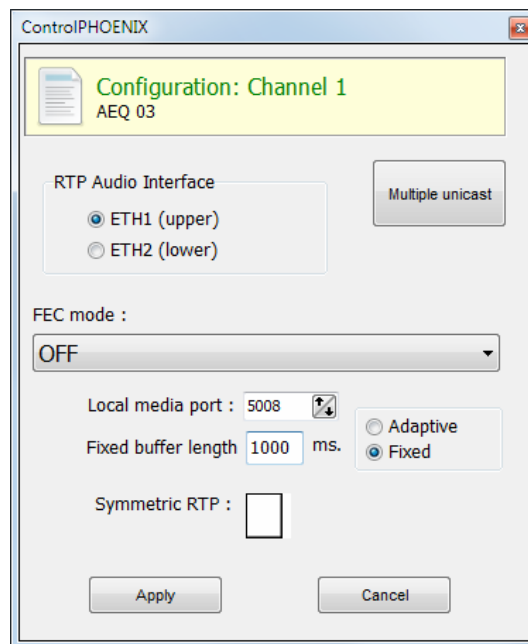


The two parameters to be configured in the dialog for any of these last four modes are:

- "Stun server": STUN server that will be used. By default, stun.aeq.es.
- "Stun port": STUN server port assigned by the administrator. By default, 3478.

6.1.4.3. IP communications interface configuration. RTP Raw mode. Broadcast / Multicast / Multiple Unicast.

When the logic IP communications interface (Channel 1 / Channel 2) of the unit is configured as "RTP Raw", the SIP and NAT Traversal adjustments described above don't apply and the connection between units will need to be done end to end (there is no call negotiation). There is also no negotiation of the coding based on existing profiles, but it needs to be defined at both sides exactly matching, unless "SmartRTP" option is activated (see section 6.1.5).



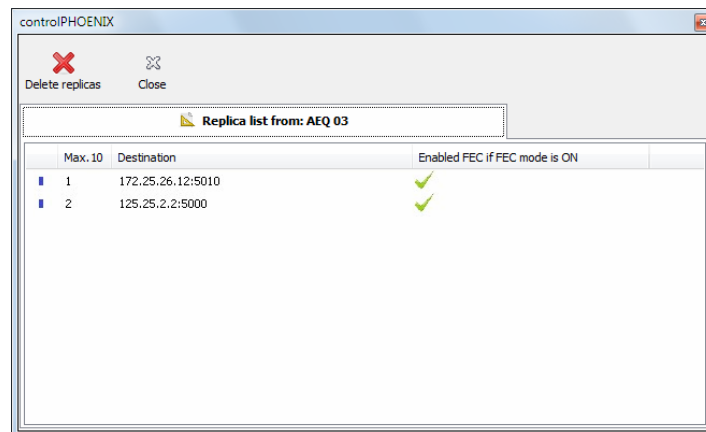
This mode allows us to send the same audio stream to several different destinations, in several ways:

a) Broadcast: the audio stream can be sent to all the equipments within a local network, only by specifying a special address in the destination address field. This address is calculated as the network address with the equipment part filled with 1's. For instance: if the IP address of our codec is 192.168.20.3 and network mask is 255.255.255.0, the corresponding broadcast address is 192.168.20.255. However, if the network mask was, for example, 255.255.0.0, then the broadcast address would be 192.168.255.255. The audio will be sent to a given port, so the receiving equipments should have "local media port" set up to this same port so they are able to receive the RTP stream. This mode is not recommended for big networks and is usually blocked by the switches and routers, so its use is restricted to small, well managed local area networks.

b) Multicast: it is also possible to send the audio stream to a special "multicast" address, for example, 239.255.20.8. If the receiving equipments call to that same IP, they will receive the audio that is being sent provided that their "local media port" matches the one the transmitter is sending the packets to. Similarly to broadcast, multicast traffic is usually blocked by switches and routers, so its use is restricted to local area networks too.

c) Multiple-unicast: Phoenix units can send the same RTP stream to several different IPs by replication of the encoded audio. This can traverse switches and routers in the same way it would do if it was a simple (unicast) RTP Raw stream, although it is limited to a certain number of destination IPs depending on the type of coding algorithm.

The **"Multiple unicast"** button gives access to an information window where the replicas that are active at a certain moment are shown.

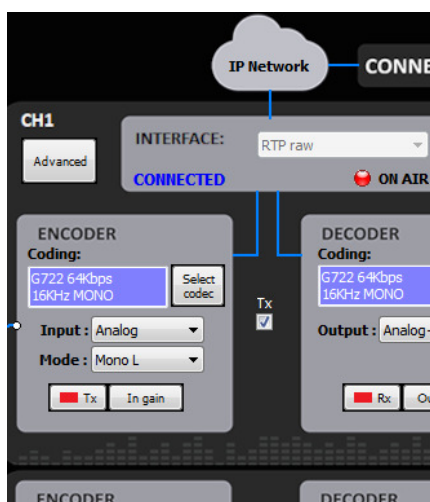


You can add replicas to a RTP call (or edit the existing ones) by previously creating a contact in the **"General"** agenda that includes them (or by editing a existing contact). See section 5.1.7.1 of this manual. The previous window allows you to delete the active replicas by means of the "Delete replicas" button (the replicas transmission stops at that very instant, during the call).

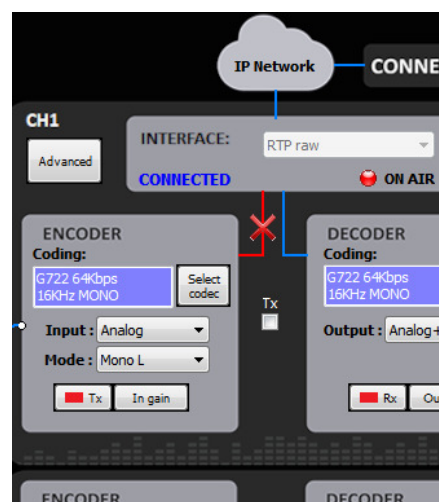
Those replicas or parallel streams are only IP+port pairs we want to send the audio to. Note that, when creating/editing the contact, FEC mode can be individually activated or disabled provided that it was enabled (and configured) in the general channel configuration.

When the previous list is empty, the stream is being sent to the IP+port indicated in the Call window.

NOTE 1: When audio is sent to several destinations, the transmitting unit can receive audio from only one of them (or none). In order to determine which one is sending audio back, only in RTP Raw mode a control will appear in the general configuration window next to the ENCODER output allowing us to disable the transmission to the IP channel. Make sure that only one of the receiving equipments is enabled to transmit:



Channel transmission activated



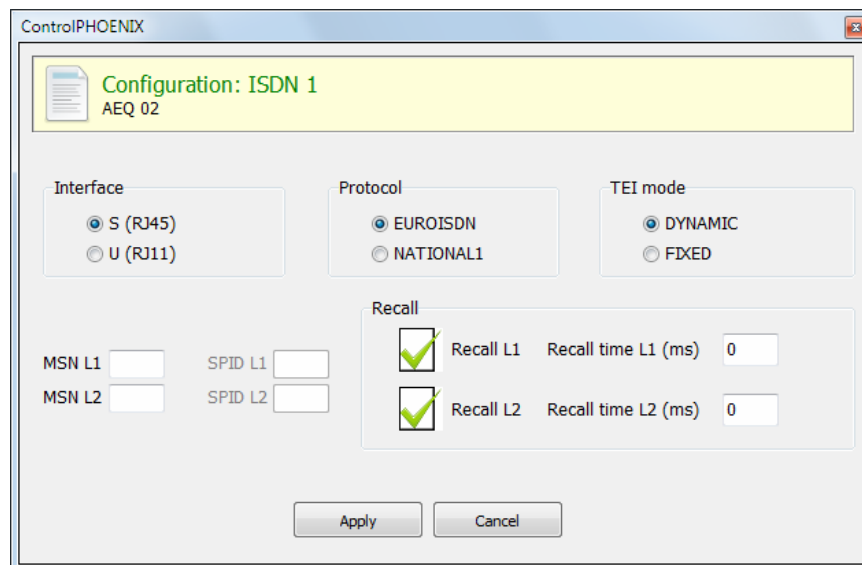
Channel transmission deactivated

NOTE 2: Advanced contacts (those allowing specification of the communication mode, coding algorithm/profile, replicas, etc) can only be stored in the General agenda, that is saved in a database in the control PC. These contacts can however be copied to the different devices, but the advanced fields (interface, coding algorithm, SIP account / provider and replicas) will be lost, so only the contact name and contact data (main IP:port or destination URI) will be stored.

NOTE 3: It is possible to use multiple-unicast transmission with "**SmartRTP**" active. The transmitter unit must be the one that generates the calls, and when it hangs up, it will send notifications to the MAIN destination only (not in the replicas). As a consequence, only this one will hang-up. If the user needs that all receivers hang-up too, the "Auto Hang-Up" option can be activated on them defining a reasonable time (let's say, 5-10 seconds). If, on the other hand, we need to cut the call from a receiver, only that with transmission enabled will make the others hang-up. In this case, the unit where the check box indicated in NOTE 1 must be activated should be the MAIN receiver.

Please read the application notes published by AEQ regarding IP connectivity for more information on IP communications in particular scenarios.

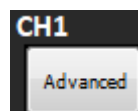
6.1.4.4. ISDN (ISDN1/ISDN2) communications interface configuration (Phoenix Stratos and Phoenix Studio only).



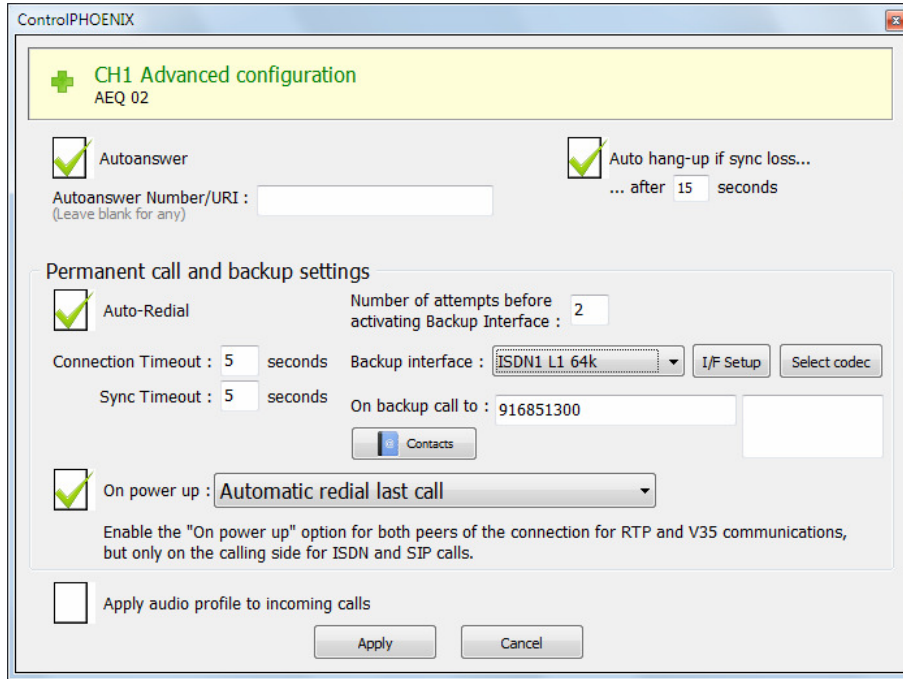
There can be up to two ISDN modules installed into **Phoenix Studio** (upper slot is ISDN1 and lower slot is ISDN2). **Phoenix Stratos** includes an integrated ISDN module.

In this case, the type of physical interface will be selected (S or U), the ISDN protocol (EuroISDN or National 1) and whether the TEI is dynamic or fixed. We will also indicate the MSN numbers (SPID if protocol is National 1), whether we want automatic recall or not and its timing.

6.1.5. "Advanced" channels configuration.



By clicking on the "Advanced" button, we reach the advanced channel configuration screen.



ControlPHOENIX

CH1 Advanced configuration
AEQ 02

☒ Autoanswer
Autoanswer Number/URI :
(Leave blank for any)

☒ Auto hang-up if sync loss...
... after 15 seconds

Permanent call and backup settings

☒ Auto-Redial
Number of attempts before activating Backup Interface : 2

Connection Timeout : 5 seconds
Sync Timeout : 5 seconds

Backup interface : ISDN1 L1 64k I/F Setup Select codec

On backup call to : 916851300

☒ On power up : Automatic redial last call

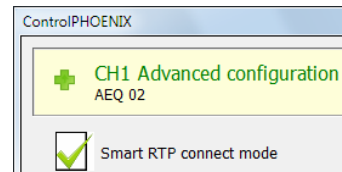
Enable the "On power up" option for both peers of the connection for RTP and V35 communications, but only on the calling side for ISDN and SIP calls.

☐ Apply audio profile to incoming calls

These are the parameters that can be configured:

- **"Autoanswer"**: allows you to activate the automatic answer of every incoming call or only those originated from a given caller.

When the operating mode for the currently selected channel has been set to RTP, this option is substituted by **"SmartRTP connect mode"**. This option allows you to activate "SmartRTP", the AEQ proprietary protocol that greatly simplifies the task of making an RTP call, so when operating Phoenix family of audiocoders with this mode activated, it is only required that the "caller" launches the call. The remote equipment will automatically answer and send its audio stream to the callers IP and port. Further, the audiocoder will also detect and automatically select the encoding algorithm that the calling unit is using to initiate the communication. The call does not need to be manually accepted and the hang-up event (from any end) will also be signaled to either unit.



ControlPHOENIX

CH1 Advanced configuration
AEQ 02

☒ Smart RTP connect mode

- **"Auto hang-up"**: automatic hang-up when audio packets are not received from IP interface after configured time. This option is not available for ISDN and V35.
- **"Permanent call and backup settings"**: allows you to activate automatic redial and configure a backup interface and define the time and number of retries before switching, the number or unit to call to and the used encoding. You can select no backup interface ("NONE") and activate the same way the **"Auto-Redial"** check box when you simply want a call to be retried in case of connection falls.

When selected mode is RTP or V35 and "Auto-Redial" option is active, when synchronism is lost the call will not be disconnected: it saves time of recovery and allows to keep the call connected when there is no return (for instance, in multiunicast links).

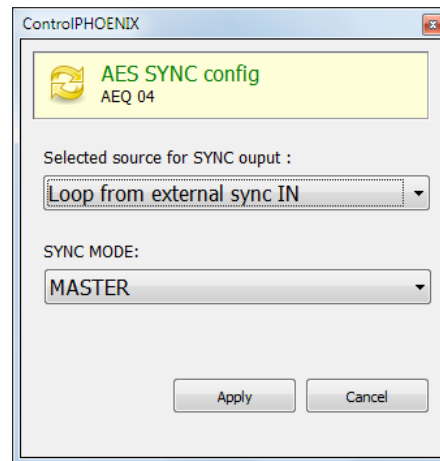
The **"On power up"** check box and the associated drop-down menu allow you to activate the automatic redial when equipment starts up. This option should be activate for both peers of the connection for RTP or V35 communications and only on the calling side for ISDN or SIP calls.

- **"Apply audio profile to incoming calls"**: allows you to filter the SIP calls depending on the encoding profile of the receiver (when any of the encoding modes included in that profile is also included in the profile of the calling unit, then communication will be established).

6.1.6. External synchronization configuration.

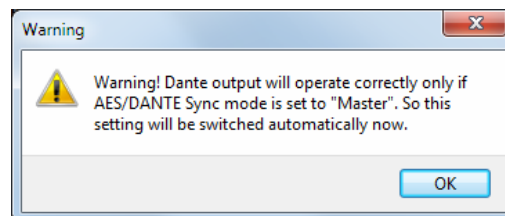
AES SYNC Config

There is a last option related with digital audio AES outputs. This button gives access to a dialog where you can configure whether the digital audio synchronization is extracted from sync input (SLAVE) or from the channel (MASTER). In the case of **Phoenix Studio**, you can also configure whether the sync output signal follows the sync input ("Loop from external sync IN") or comes from the channel extracted sync.

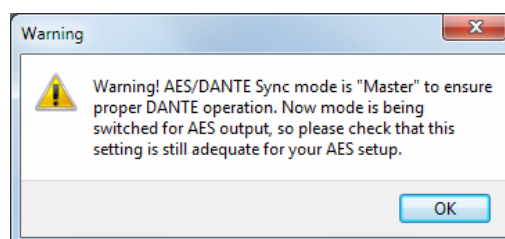


IMPORTANT NOTE: In the case of **Phoenix Venus V3, V4** and **V4+**, if you use the Dante AoIP outputs, "SYNC MODE" must be configured as "MASTER". From version **2.2.2.9** of the application on, that configuration is accomplished automatically when the "Analog+AoIP" option is selected for any of the outputs (an informative window appears):

AES/DANTE SYNC Config

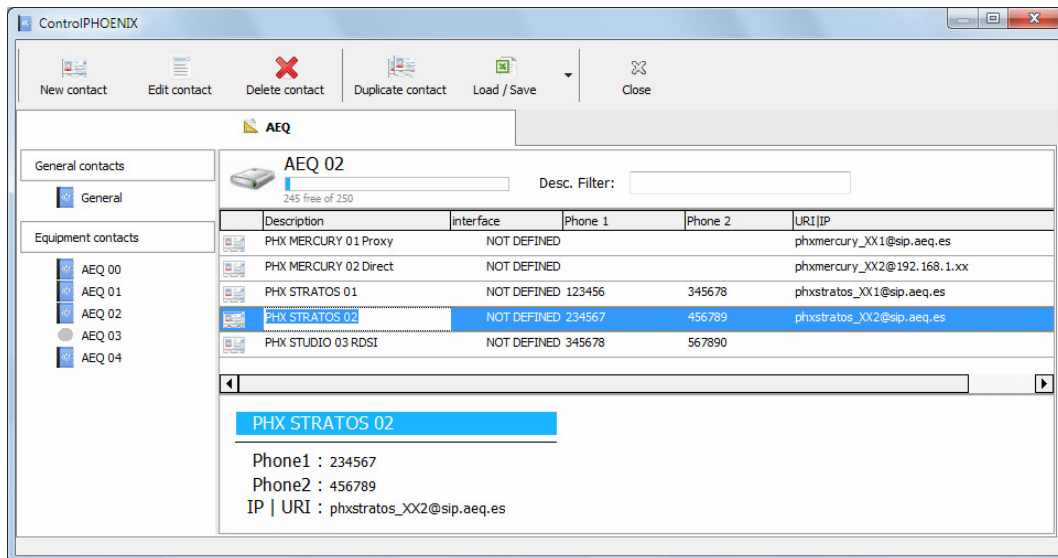


If then you return to the "Analog+Digital" option, "SYNC MODE" configuration as "MASTER" remains (an informative window also appears):



6.2. Contacts window.

This option gives access to the contacts window associated to the selected codec. See section 5.1.7.



6.3. Ethernet window.

This option allows you to manage the configuration of the IP parameters of the unit Ethernet interface (2 interfaces in the case of **Phoenix Studio**, **Phoenix Venus V2**, **Phoenix Venus V3**, **Phoenix Venus V4** and **Phoenix Venus V4+**).

The parameters to be configured for each interface are:

- **"Enable DHCP"**: enables the activation or deactivation of the automatic configuration of IP addresses, masks and gateways. There must be a DHCP server in the network the unit is connected to in order to make this option work. When "Enable DHCP" is validated, the following parameters will be filled automatically; when "Enable DHCP" is not validated you will be able to change them manually.
- **"IP Address"**: valid IP address associated with that interface.
- **"Subnet mask"**: valid subnet mask associated with that interface.
- **"Gateway IP"**: valid gateway or network gateway address associated with that interface. The **"Default Gateway"** checkboxes (mutually exclusive) allow to select which gateway (the one configured for interface 1 or the one configured for interface 2) will be used for incoming control & management communications outside the defined local networks; note that the interface used for SIP/RTP communications is explicitly defined in the corresponding windows (see section 6.1.4.2 of this manual), so this setting has no effect on them.
- **"DNS Server"**: IP address of the external addresses resolution server, valid in the geographic zone where codec is placed, or internal server (inside the local network) authorized to translate alphanumeric URL identifiers into IP addresses.

ControlPHOENIX


Ethernet config
AEQ 03

Ethernet module 1
OK

☐ Enable DHCP

IP Address
192 . 168 . 1 . 86

Subnet mask
255 . 255 . 255 . 0

Gateway IP
192 . 168 . 1 . 100
☒ Default Gateway*

DNS Server
8 . 8 . 8 . 8

Ethernet module 2
NO_LINK

☐ Enable DHCP

IP Address
172 . 26 . 33 . 80

Subnet mask
255 . 255 . 0 . 0

Gateway IP
172 . 26 . 1 . 1
☐ Default Gateway*

DNS Server
8 . 8 . 4 . 4

*Select which Gateway (ETH1's or ETH2's) will be used for incoming control & management communications outside the defined local networks. NOTE that the interface used for SIP/RTP communications is explicitly defined (RTP interface and SIP provider interface), so this setting has no effect on them.

for changes to apply, the equipment needs to reboot. Communication will be re-established in aprox. 15 sec.

Apply
Cancel

Once those parameters are configured, and after pressing the "Apply" button, a confirmation window will appear. After confirming, the equipment reboots and the communication re-establishes in approximately 15 seconds.

ControlPHOENIX



Warning: An incorrect configuration of the network parameters can lead to loss of communication with the equipment.

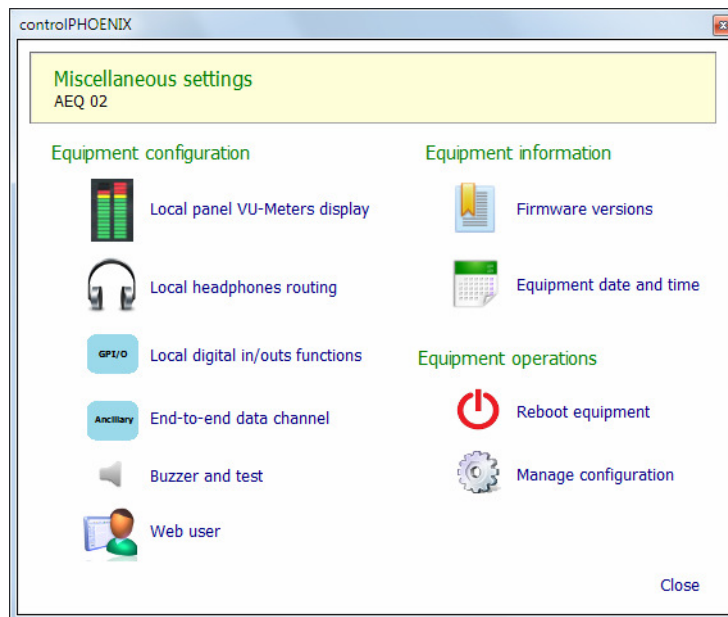
For changes to apply, the equipment needs to reboot. Communication will be re-established in aproxiamaty 15 sec.

Do you really want to continue ?

Yes
No

6.4. Miscellaneous window.

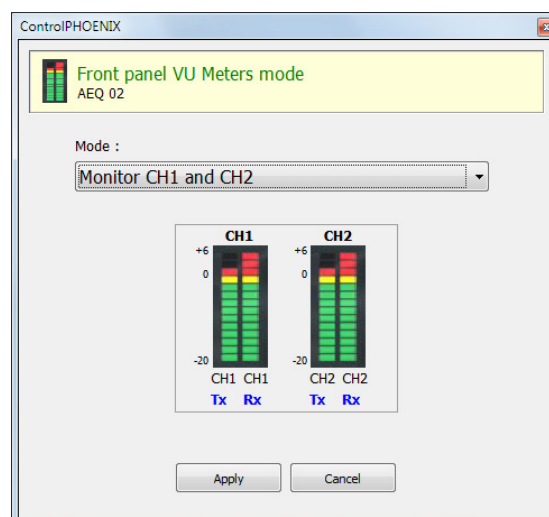
Allows the user to configure certain normally permanent details in the equipment.



6.4.1. Local panel VU-Meters display (Phoenix Stratos and Phoenix Studio only).

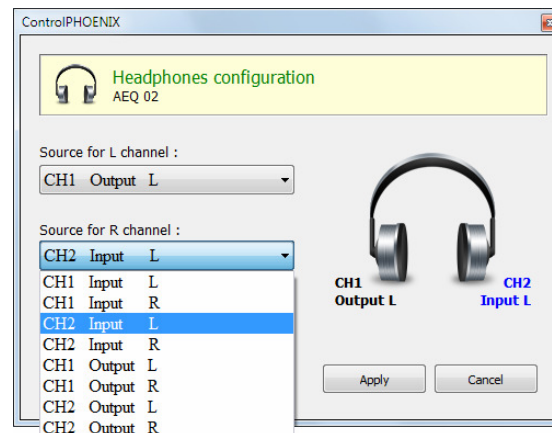
This option allows the user to configure the signals to monitor in the front panel display of the equipment, among three different options:

- Inputs and outputs of channel 1 in stereo.
- Inputs and outputs of channel 2 in stereo.
- Inputs and outputs of both channels at the same time (but displaying only "L" input/output of channel 1 and "R" input/output of channel 2).



6.4.2. Local headphones routing (Phoenix Stratos and Phoenix Studio only).

This option allows the user to select the signals to be monitored in each channel of the headphone connected to the front panel jack, among 8 different sources.

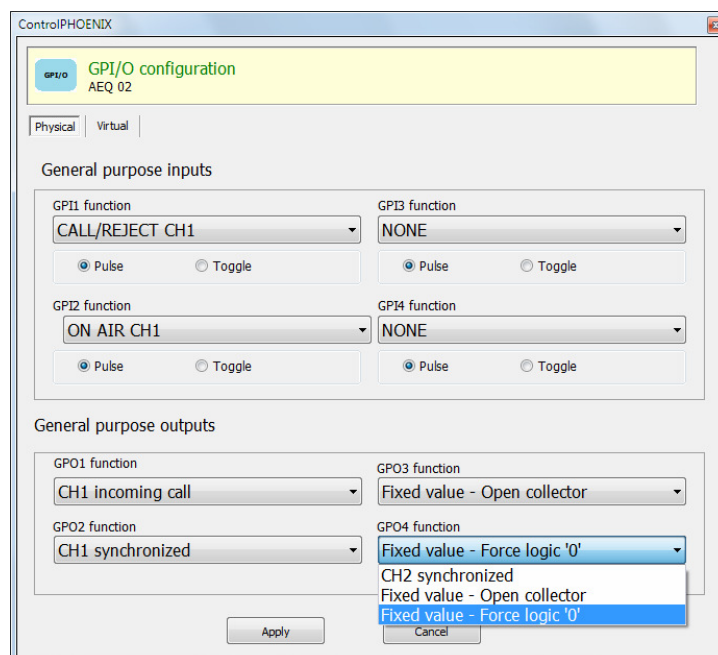


6.4.3. GPI/O - Local digital in/outs functions (Phoenix Stratos, Studio, Venus, Venus V2, Venus V3, Venus V4 and Venus V4+ only).

This option allows the user to configure the **physical** and **virtual** GPI/Os of the unit, through the "Physical" and "Virtual" tabs of the window associated to this option.

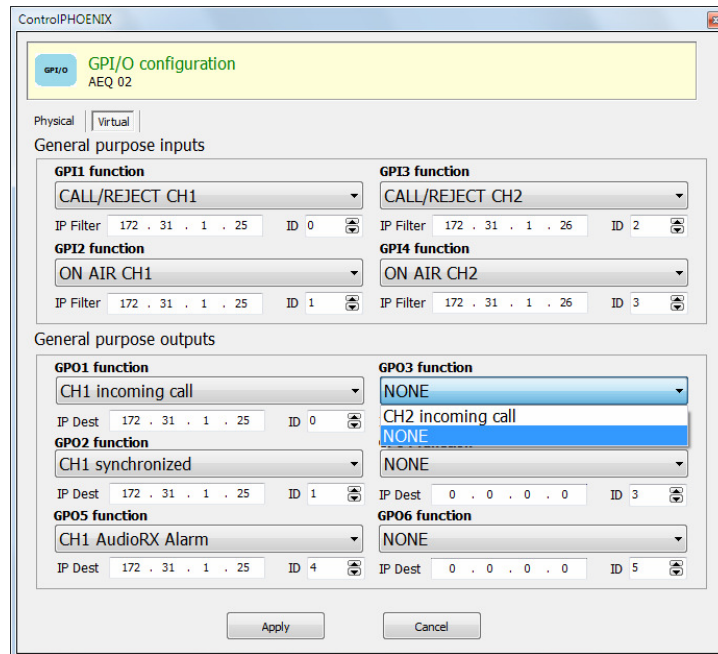
Regarding the **physical GPI/Os**, **Phoenix Studio** counts with 2 inputs and 2 outputs, one pair assigned to each channel. **Phoenix Stratos, Venus, Venus V2, Venus V3, Venus V4 and Venus V4+** have 4 inputs and 4 outputs (6 outputs for V3, V4 and V4+), two pairs assigned to each channel.

The **physical GPIs** are configured in "General purpose inputs" area, they can operate as pulse or by level ("Toggle"), and their function can be control of "ON-AIR" or call establishment / rejection ("CALL/REJECT"). The **physical GPOs** are configured in "General purpose outputs" area and they can indicate an incoming call, synchronized call, or fixed logic "0" or "1" values.

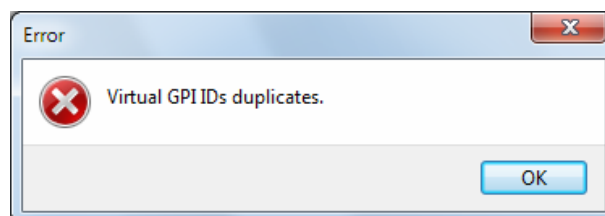


Regarding the **virtual GPI/Os**, **Phoenix Studio**, **Stratos**, **Venus**, **Venus V2**, **Venus V3**, **Venus V4** and **Venus V4+** have 4 virtual GPIs and 6 virtual GPOs (2 GPIs and 3 GPOs per channel).

"General purpose inputs" area allows you to configure the actions to execute when a **virtual GPI** is received: put the channel "ON-AIR" or call establishment / rejection ("CALL/REJECT").

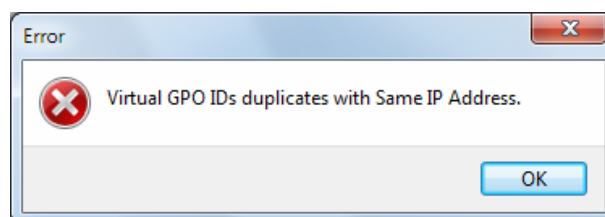


For each one, you must define a unique identifier ("ID") and an IP address as origin of the received orders ("IP Filter"). The "ID" parameter must be not repeated for other GPIs; when you try to duplicate it, an error message will be displayed:



On the other hand, "General purpose outputs" area allows you to configure that a certain **virtual GPO** is sent when a certain system event appears. These events will be received as virtual GPIs in the destination unit. There are three events types that can cause that a virtual GPO is sent from a PHOENIX audiocodec to other unit: incoming call (for Proxy/Direct SIP modes), synchronized call (also for RTP mode) and audio Rx alarm (according to configured threshold).

For each one, you must define an identifier ("ID") and a destination IP address ("IP Dest"). The "ID" parameter must be not repeated for other GPOs with the same IP address; when you try to duplicate it, an error message will be displayed:

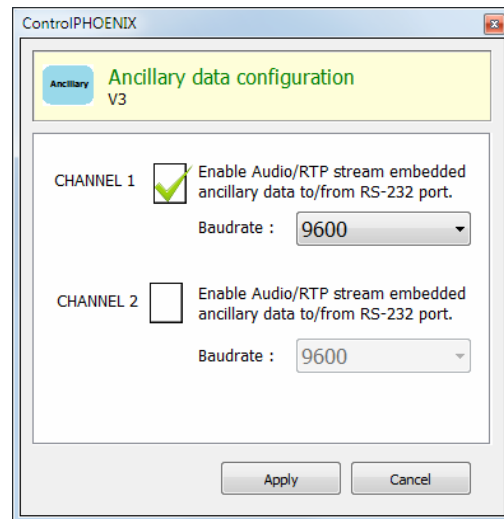
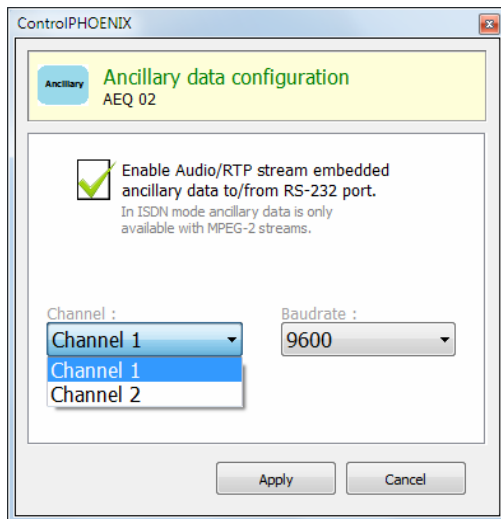


NOTE 1: According to system design, the **virtual GPIs** and **GPOs** are always sent from unit to another (virtual GPOs can not be distributed to several destinations and a certain action can not be executed when several virtual GPIs are received from different origins).

NOTE 2: The units with two network interfaces (Phoenix Studio, Venus V2, Venus V3, Venus V4 and Venus V4+) will be able to send and receive virtual GPI/Os only through **LAN1** (or **NET1**) port.

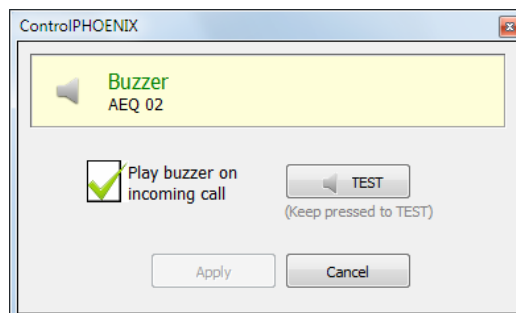
6.4.4. Ancillary - End-to-end data channel.

Sets up the ancillary data channel, allowing its activation associated to either channel 1 or 2, and selection of the binary rate that is limited to 9600 bauds in the case of ISDN transmitted channels (in Phoenix Stratos and Phoenix Studio) and can be selected between 1200, 2400, 4800, 9600, 19200 and 38400 bauds for IP connections. In the particular case of Phoenix Venus V3, V4 and V4+, there are two independent RS232 ports, each one devoted to one audio channel: both of them can be independently enabled or disabled.



6.4.5. Buzzer and test.

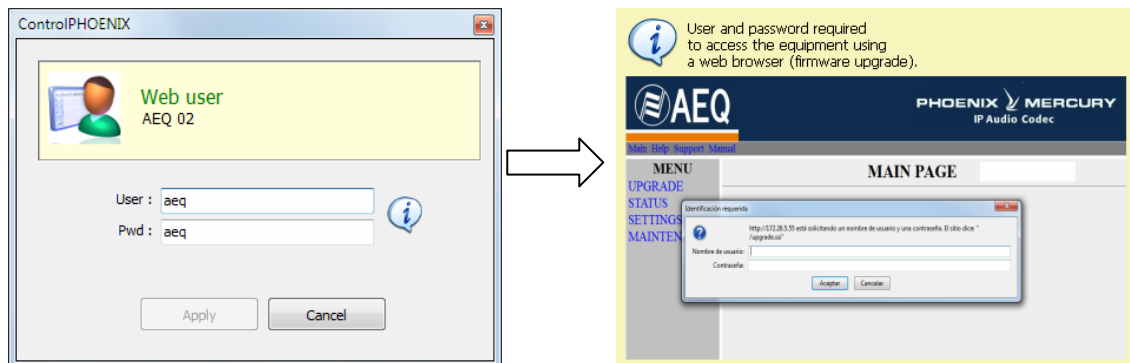
Activates, deactivates and tests the incoming call buzzer.



6.4.6. Web user.

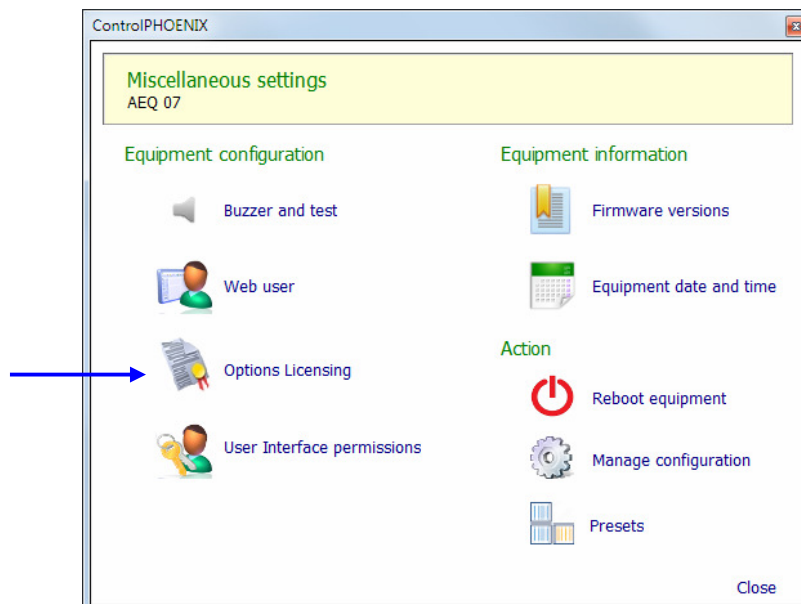
To be accessed only by trained maintenance personnel. Allows for the modification of the username & password that are requested for firmware upgrade using the web interface.

When the mouse cursor is over the  image, the following help screen will automatically appear, warning the user that this information is required to access with the web interface to the firmware upgrade procedure. This process must be done, if necessary, with the greatest care and following AEQ Technical Service request (AEQ will assume no responsibility for damages caused after improperly updating the unit's firmware if this has been done without proper care and after Technical Assistance Service's request).



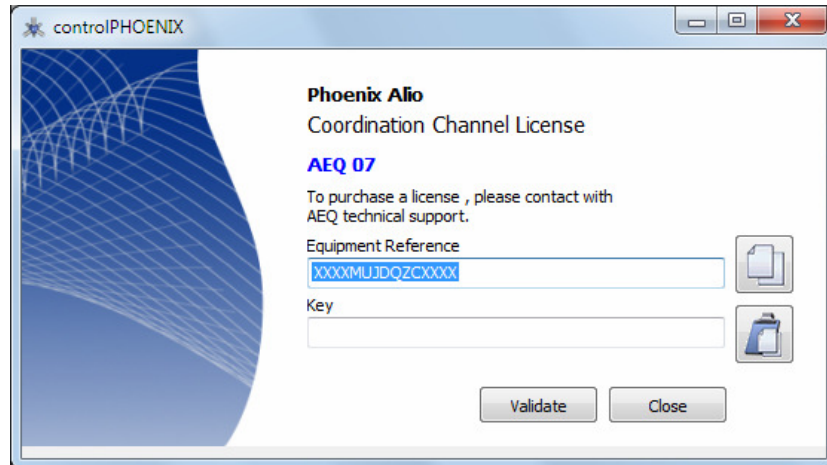
6.4.7. Options Licensing (Phoenix ALIO and Phoenix Venus V3, V4 & V4+ only).

This option allows you to activate the **COORDINATION** channel (in the case of Phoenix ALIO) or the **DANTE AoIP** interface (in the case of Phoenix Venus V3, V4 and V4+), if the corresponding **license** has been purchased.

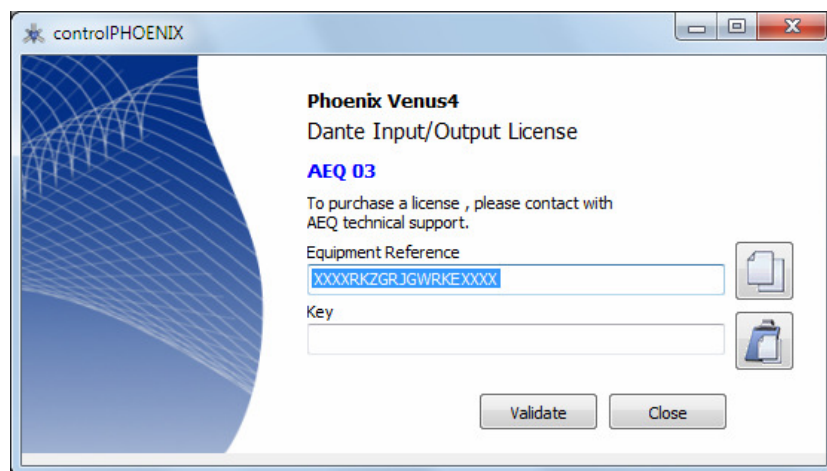


When you press "**Options Licensing**", the window where the required option can be activated appears.

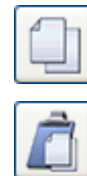
In the case of Phoenix ALIO:



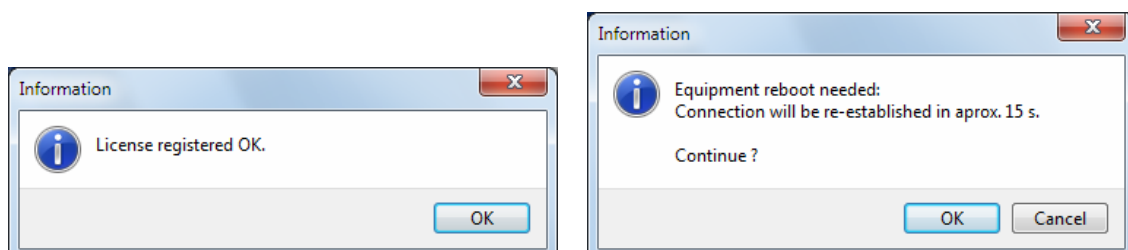
In the case of Phoenix Venus V3, V4 and V4+:



In case your order includes any of those options, the unit is supplied from Factory with the option activated. If you want to activate it a posteriori, you must copy the **"Equipment Reference"** by means of the associated button and send it to AEQ in order to generate the corresponding **"Key"** and send it back to you (you must paste it by means of the associated button).

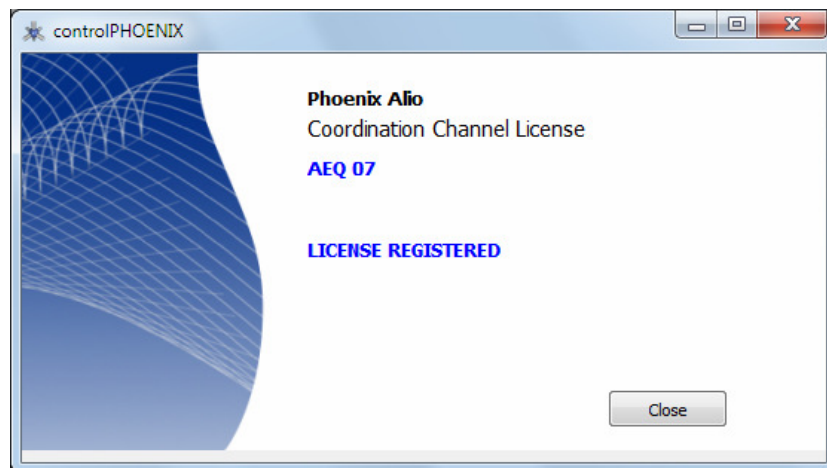


The **"Validate"** button allows you to confirm the key and, when it's correct, activate the option. Then two confirmation windows will appear in order to inform that license is correctly registered and that an equipment reboot is needed.

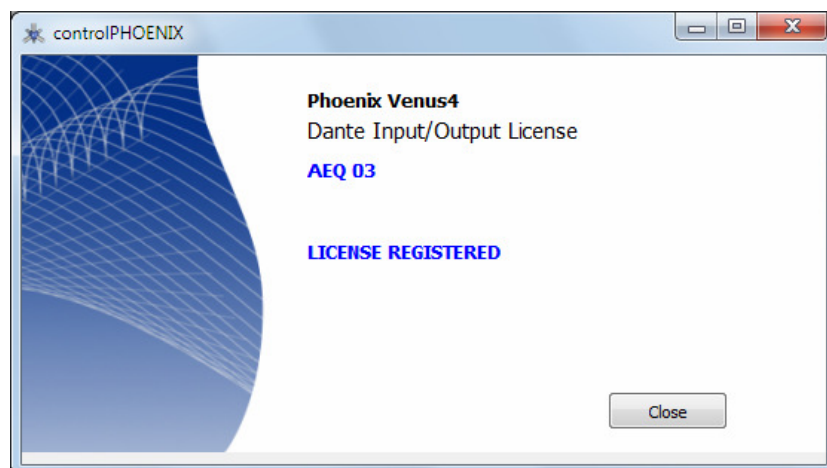


After rebooting, if you press again "Options Licensing" the window will show that license is registered.

In the case of Phoenix ALIO:



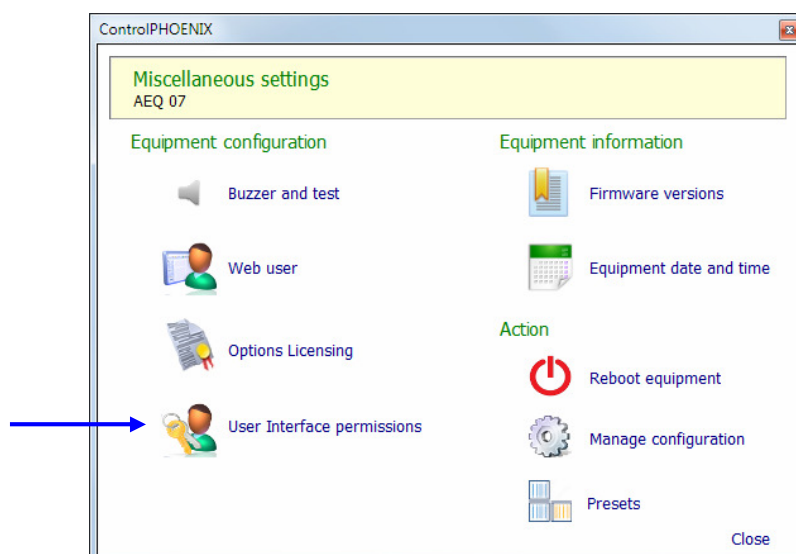
In the case of Phoenix Venus V3, V4 and V4+:



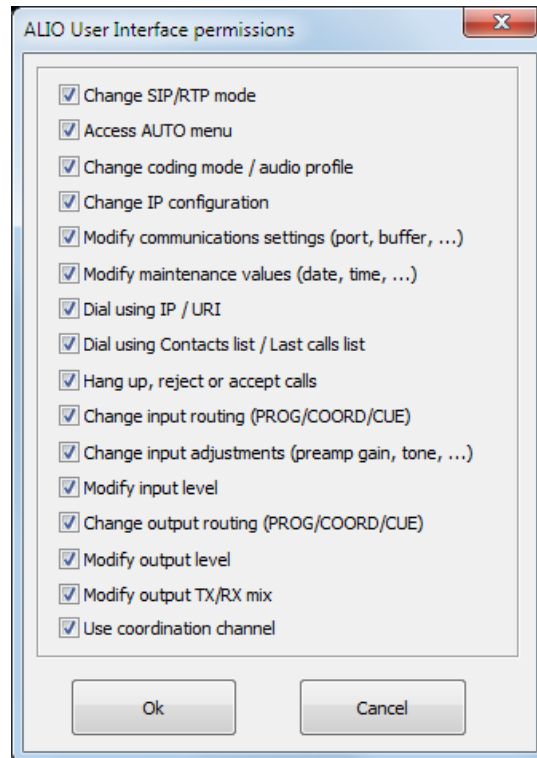
If you have any doubt, please consult the AEQ Technical Service (support@aeq.es) or authorized distributors.

6.4.8. User Interface permissions (Phoenix ALIO only).

This option allows you to configure different user interface permissions for Phoenix ALIO.



When you press **"User Interface permissions"**, a window will appear that allows you to configure those permissions:



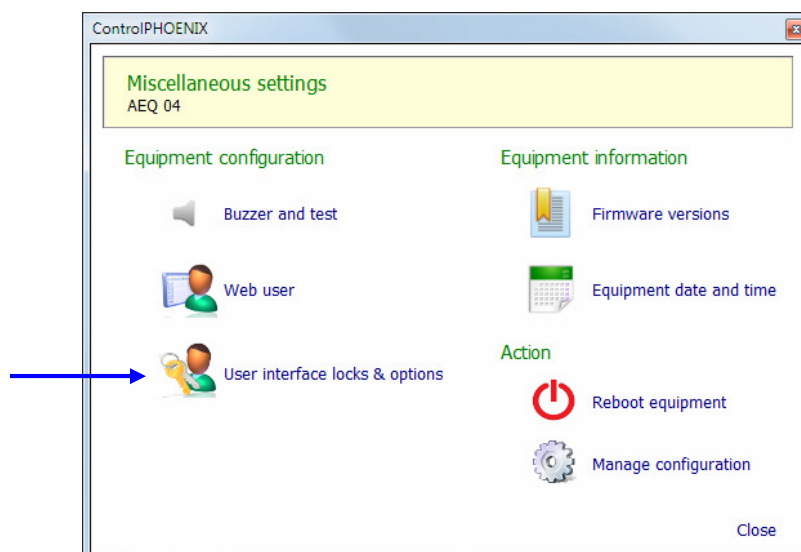
All the permissions are active by default.

In order to deactivate any of those permissions, locking the associated function of the unit's user interface, you just have to deselect the corresponding checkbox and press "Ok". In order to reactivate any of them, you just have to select the corresponding checkbox and press "Ok". Only for the last permission, you also have to turn the device off and on after pressing "Ok".

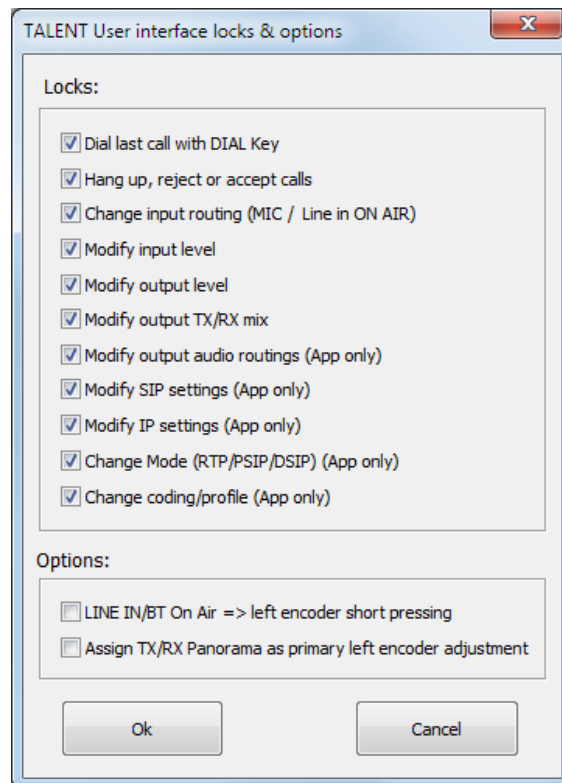
For more information, please consult the Phoenix ALIO user's manual (section 4.9).

6.4.9. User interface locks & options (TALENT only).

This option allows you to configure different user interface permissions for TALENT.



When you press **"User interface locks & options"**, a window will appear that allows you to configure those permissions:



By default, all the "Locks" are activated and the 2 "Options" are deactivated.

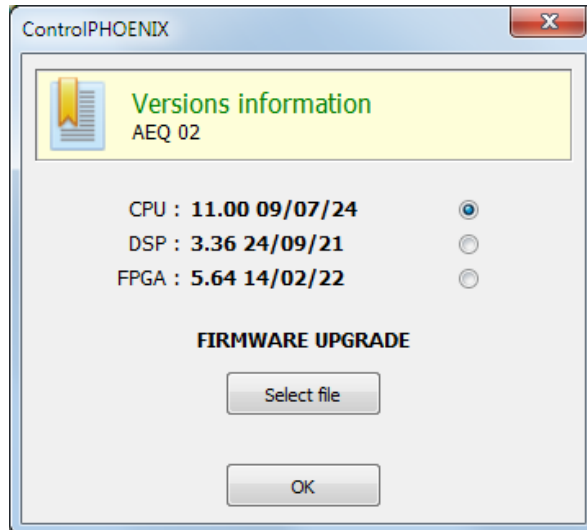
In order to deactivate any of those "Locks", locking the associated user interface function, you just have to deselect the corresponding checkbox and press "Ok". In order to reactivate any of them, you just have to select the corresponding checkbox and press "Ok".

In order to activate any of those "Options", modifying the associated user interface function, you just have to select the corresponding checkbox and press "Ok". In order to deactivate any of them, you just have to deselect the corresponding checkbox and press "Ok".

For more information, please consult the TALENT user's manual (section 4.9).

6.4.10. Firmware versions.

This option shows the firmware versions installed in the equipment for each one of the programmable elements of the system: CPU, DSP and FPGA for Phoenix ALIO, Phoenix Stratos and Phoenix Studio (only CPU and DSP for TALEN, Phoenix Mercury, Venus, Venus V2, Venus V3, Venus V4 and Venus V4+). Each version has particular technical features, so this information is mandatory for any query to AEQ Technical Service.

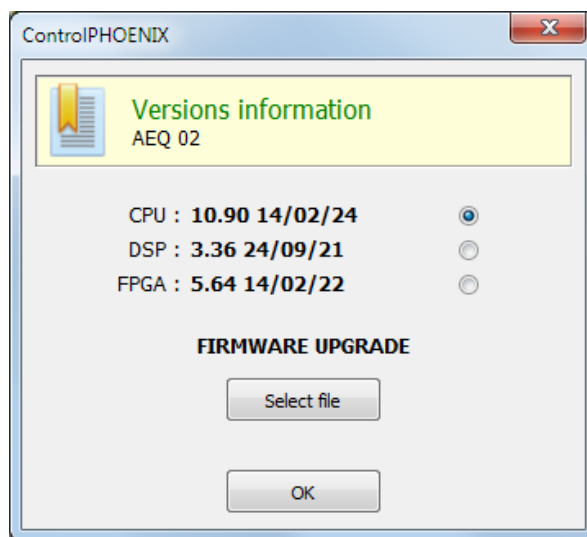


IMPORTANT NOTE: Please, don't upgrade the equipment firmware without consulting AEQ Technical Service previously.

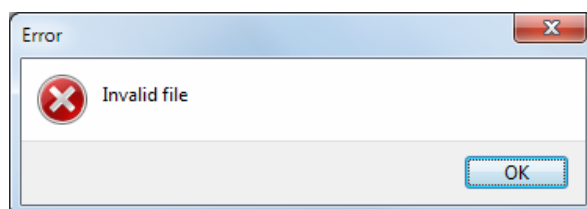
6.4.10.1. Firmware Upgrade.

From the previous window it is also possible to accomplish the firmware upgrade. First you must select the module you want to upgrade (CPU in the example).

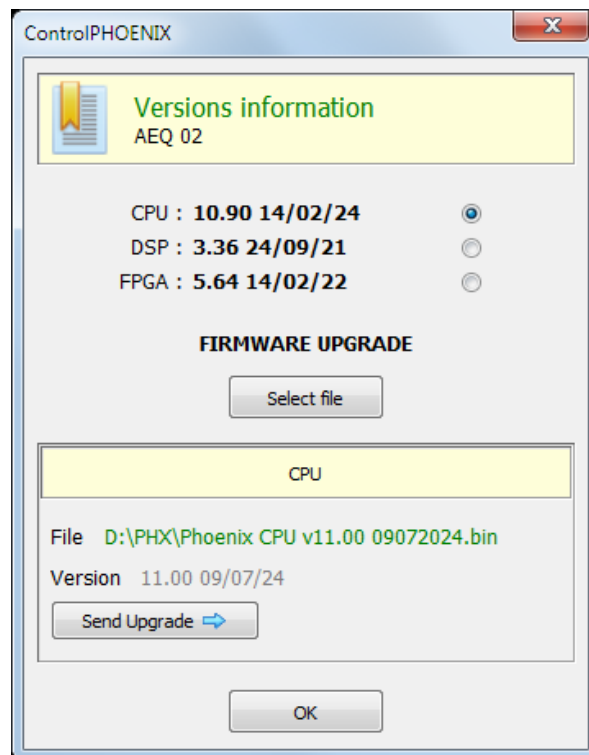
IMPORTANT NOTE: The recommended order for upgrading is: CPU, DSP and FPGA.



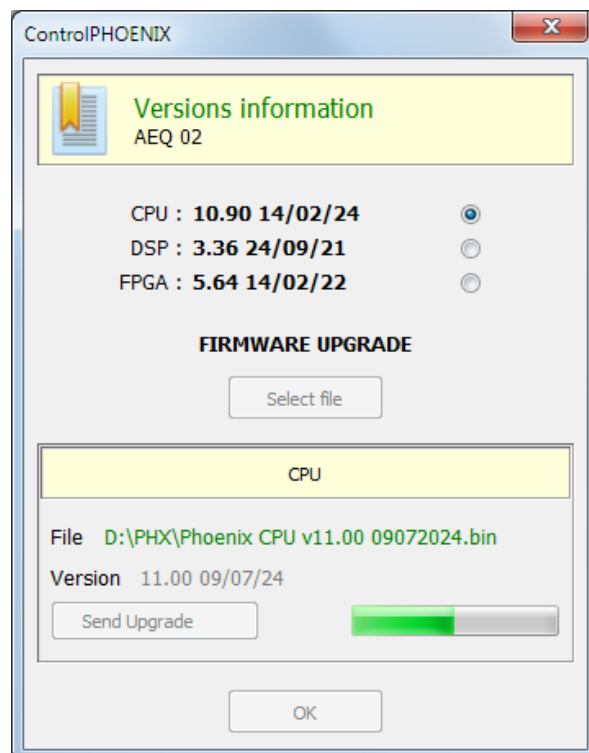
Next, press the "Select file" button and, in the window that appears, select the upgrade file containing the new firmware version. In case you select a file that does not correspond to the selected module, an error message will be displayed:



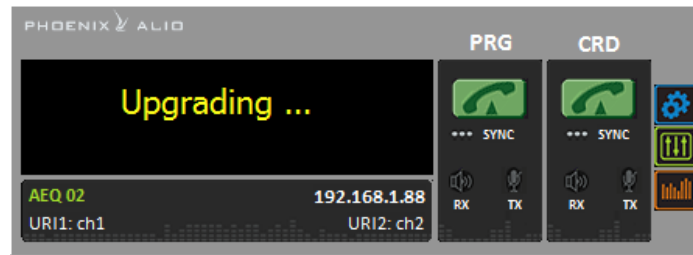
If the file is correct the window changes to:



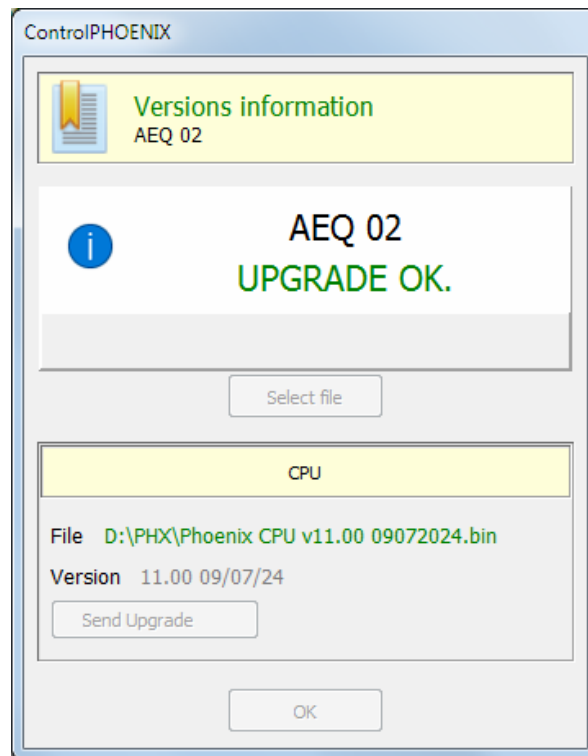
Then press the "Send Upgrade" button to start the upgrading process.



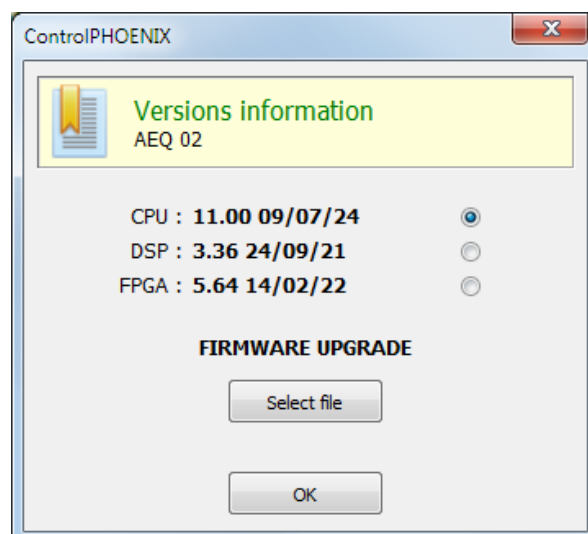
During the process, the equipment will show the following indication:



Once the process is finished, the following confirmation appears:

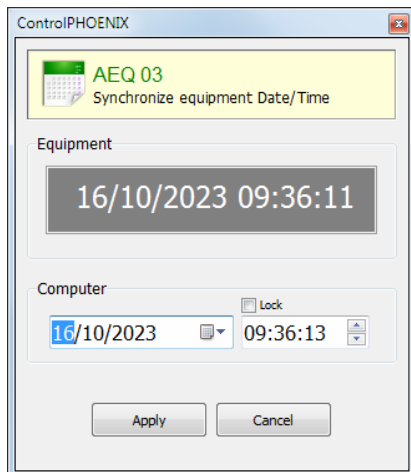


Now return to the "Firmware versions" option and ensure that the new firmware version appears.



IMPORTANT NOTE: The firmware upgrade can also be accomplished from the unit's **WebServer** (please consult section "Upgrading the equipment firmware" of the corresponding audiocodec manual).

6.4.11. Equipment date and time.

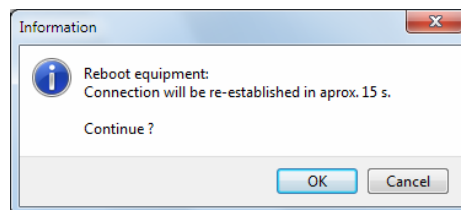


Allows you to consult equipment date/time and synchronize it with the computer's date/time. All codecs in the installation and computers controlling them should keep the same time.

6.4.12. Reboot Equipment.

Remote equipment rebooting may be necessary in certain occasions.

When you press the  **Reboot equipment** button, a confirmation window will appear:

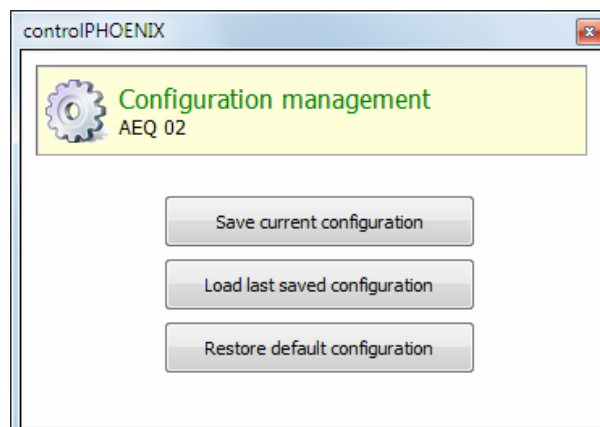


After acceptance, the equipment resets and, after around 15 seconds, the connection will be recovered. This situation is indicated by a blinking "Disconnected" legend in the equipment's main screen and a red band in the corresponding line of the equipments list view.

6.4.13. Manage configuration.

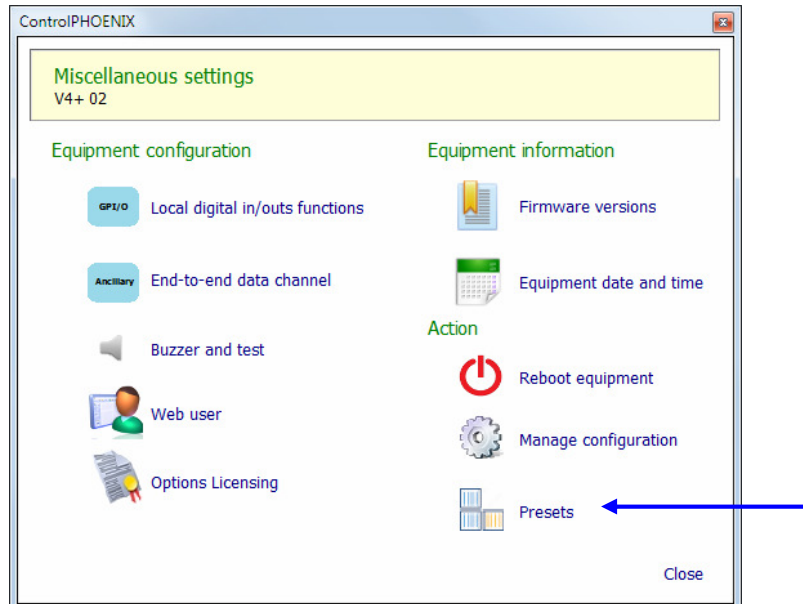
It is possible to store the current equipment configuration into its own internal non-volatile memory. It may be very useful to do this once the unit is correctly configured and working well in the intended application, so if later on someone inadvertently changes something that prevents normal operation, it is possible to recover the last stored configuration.

Besides, an additional option for restoring the factory settings is available (except for the IP settings, which are kept in order to avoid losing connectivity with the unit).

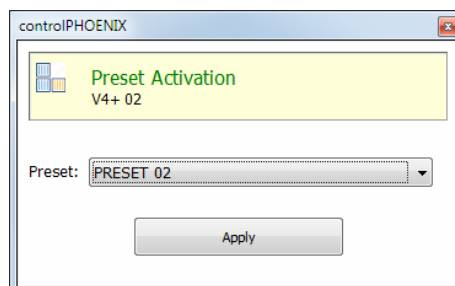


6.4.14. Presets (Phoenix Venus V4, V4+ and ALIO only).

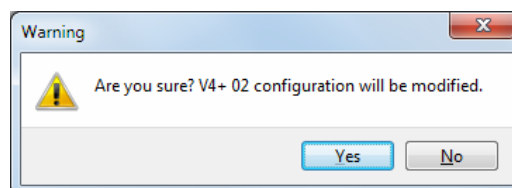
This option allows you to access the list of PRESETs saved on the unit in order to select and activate any of them. For more information, please consult section "5.1.8. Presets" of this manual.



When you press "**Presets**", the screen that allows you to make that selection appears:



Once the desired PRESET is selected, press "Apply": confirmation is requested.



If you don't wish to activate the PRESET, it's possible to deny this activation and then close the previous window using the upper right cross.

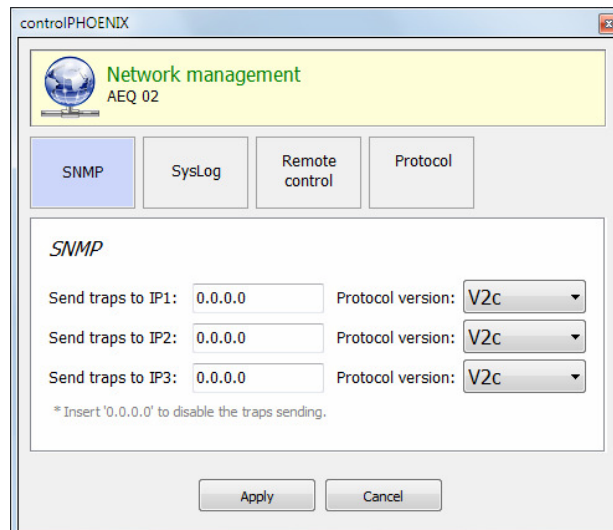
In the case of Phoenix Venus V4+ and ALIO you can also access the list of PRESETs and activate any of them from the front panel of the unit.

6.5. Network window.

Network button, placed within the Configuration menu, gives access to the "**Network management**" menu, where you can establish how the equipment behaves in what relates to remote management, that includes four submenus.

6.5.1. SNMP.

AEQ Phoenix audiocodecs family are equipped with a server able to send messages to up to 3 different IPs according to SNMP (Simple Network Management Protocol), that is very useful for remote management of complex installations consisting on many different pieces of equipment from various manufacturers.



The screenshot shows the 'controlPHOENIX' window with the 'Network management' title bar. Below the title bar is a yellow header with a globe icon and the text 'Network management AEQ 02'. There are four tabs: 'SNMP' (selected), 'SysLog', 'Remote control', and 'Protocol'. The 'SNMP' tab contains the following fields:

- Send traps to IP1: 0.0.0.0
- Protocol version: V2c
- Send traps to IP2: 0.0.0.0
- Protocol version: V2c
- Send traps to IP3: 0.0.0.0
- Protocol version: V2c

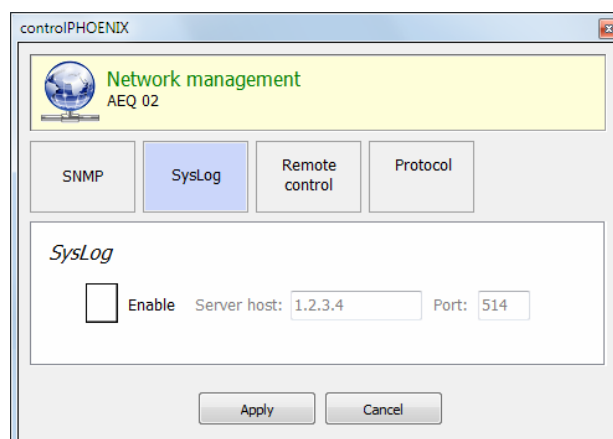
Below these fields is a note: "* Insert '0.0.0.0' to disable the traps sending." At the bottom are 'Apply' and 'Cancel' buttons.

The "**SNMP**" tab within "**Network management**" window allows you to define those three IP addresses (filling all with "0.0.0.0" is equivalent to disable the transmission of SNMP messages), as well as the version of the protocol to be used with each remote client (v1 or v2c), in order to guarantee the interoperability with the widest variety of clients so the user can choose those that better fit his needs, even when we are adding management of Phoenix audiocodecs to a previously deployed installation.

The units can send status information messages, traps or alerts, and receive simple configurations usually related to these traps. In order to get more information about the particular alerts provided by each particular model, please check its corresponding manual.

6.5.2. SysLog.

Phoenix family members can send detailed debug messages in the form of industry de-facto SysLog format in order to provide best information to our Technical Service. These messages will be sent to a configured remote client.



The screenshot shows the 'controlPHOENIX' window with the 'Network management' title bar. Below the title bar is a yellow header with a globe icon and the text 'Network management AEQ 02'. There are four tabs: 'SNMP', 'SysLog' (selected), 'Remote control', and 'Protocol'. The 'SysLog' tab contains the following fields:

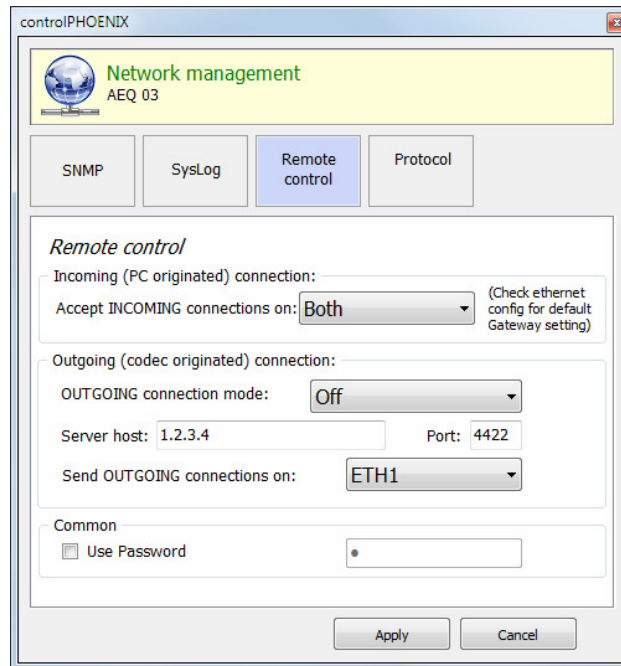
- ☐ Enable
- Server host: 1.2.3.4
- Port: 514

At the bottom are 'Apply' and 'Cancel' buttons.

The **"SysLog"** tab within **"Network management"** allows us to enable the emission of SysLog traces as well as to define the IP of the machine where the client is installed into, and the port it is listening to.

IMPORTANT NOTE: SysLog traces are disabled by default and it must be enabled only after request of AEQ Technical Service following an incidence. They will provide the necessary data to fill this dialog.

6.5.3. Remote control.



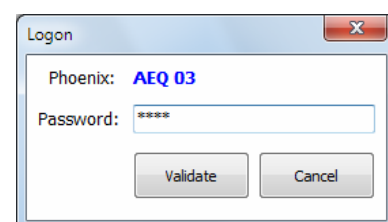
The screenshot shows the 'controlPHOENIX' application window with the 'Network management' tab selected for 'AEQ 03'. The 'Remote control' sub-tab is active. It contains settings for incoming and outgoing connections, a server host, port, and a common password section.

This option allows you to configure several parameters related to the way the equipment connects to the software application described in this manual. In **"Incoming (PC originated) connection"** section you can define which physical ports (ETH1, ETH2, both of them or none) the unit is listening remote control commands at, using the "Accept INCOMING connections on" drop-down menu. Phoenix Studio, Phoenix Venus V2, Phoenix Venus V3, Phoenix Venus V4 and Phoenix Venus V4+ have two ports, while the rest of Phoenix family members provide only ETH1 port. For the units with two ports you can define through which of them connections from an external network can be received by means of the "Default Gateway" option of Ethernet window (see section 6.3 of this manual).

Furthermore, in **"Outgoing (codec originated) connection"** section, if we select "Automatic" for the "OUTGOING connection mode", the unit will repeatedly try to connect to the indicated Server Host IP and port. This eases the control of pieces of equipment connected behind firewalls, etc. The physical interface to be used by this auto-connection mode is defined in the "Send OUTGOING connections on" drop-down menu.

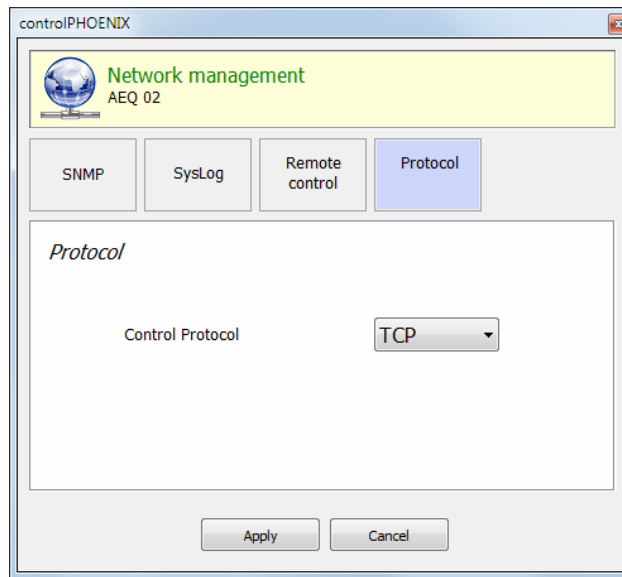
Finally, in **"Common"** section you can also configure for that unit a numerical **password** of up to 4 digits by clicking on "Use Password" checkbox.

That password will be requested whenever you try to **add** the unit to the application, whether it's done by autodiscovery or manually (once added, the password won't be requested again although you close and reopen the application, but it will be requested if you delete the unit and try to add it again or if you add it from another PC where the application is also installed). When you enter a wrong password an error message will be displayed.



The screenshot shows a 'Logon' dialog box with fields for 'Phoenix: AEQ 03' and 'Password: ****'. There are 'Validate' and 'Cancel' buttons at the bottom.

6.5.4. Protocol.

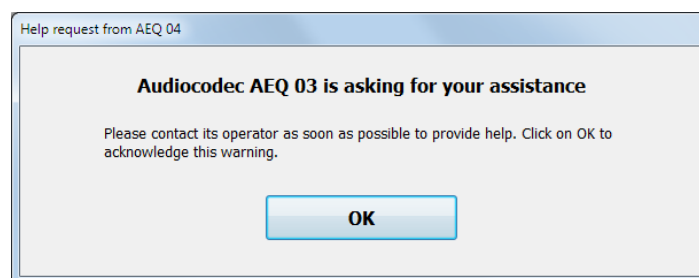


This option allows you to select the control protocol. The available options are TCP and UDP. This functionality will be available in future firmware and software versions. Any change in this option must always be made under the supervision of AEQ Technical Service (support@aeq.es).

6.6. Request for help from Phoenix ALIO / TALENT ("HELP" key).

Both Phoenix ALIO and TALENT have a key named "HELP" on their physical interface. This key allows you to send a notification to the remote control software provided that this is in use. This way, a Phoenix ALIO or TALENT user that requires assistance or has doubts can ask the operator that is remotely controlling or monitoring the unit with the "AEQ ControlPHOENIX" software for help.

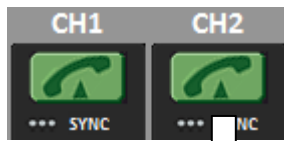
When the "HELP" key is pressed in the unit, the key will start to flash in red. On the remote control software a pop-up notification will appear asking the operator to get in touch with the user of the unit.



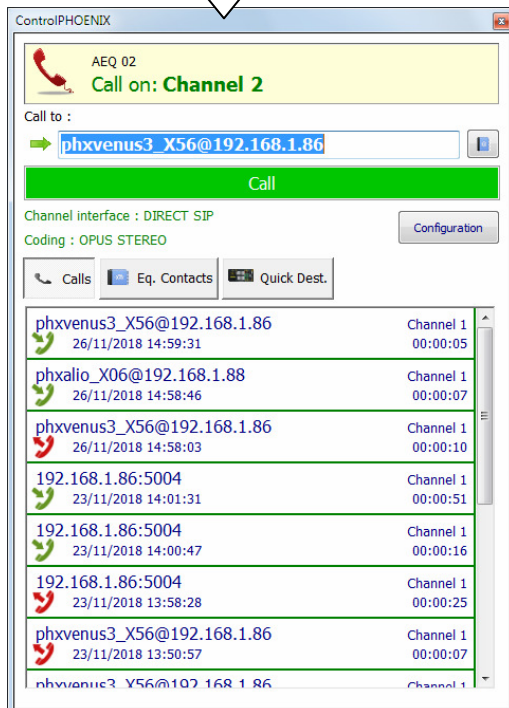
Once this notification is confirmed by the remote operator (by clicking on the "OK" button), this notification will disappear and the unit's key will stop flashing.

The notification can be cancelled from the unit at any moment by pressing the "HELP" key again.

6.7. CALL button: establishing a call.



The individual codec control window of Phoenix ALIO, Phoenix Stratos, Phoenix Studio and Phoenix Venus has a CALL button per channel. Phoenix Mercury and TALENT have only one channel so only one call button is shown.

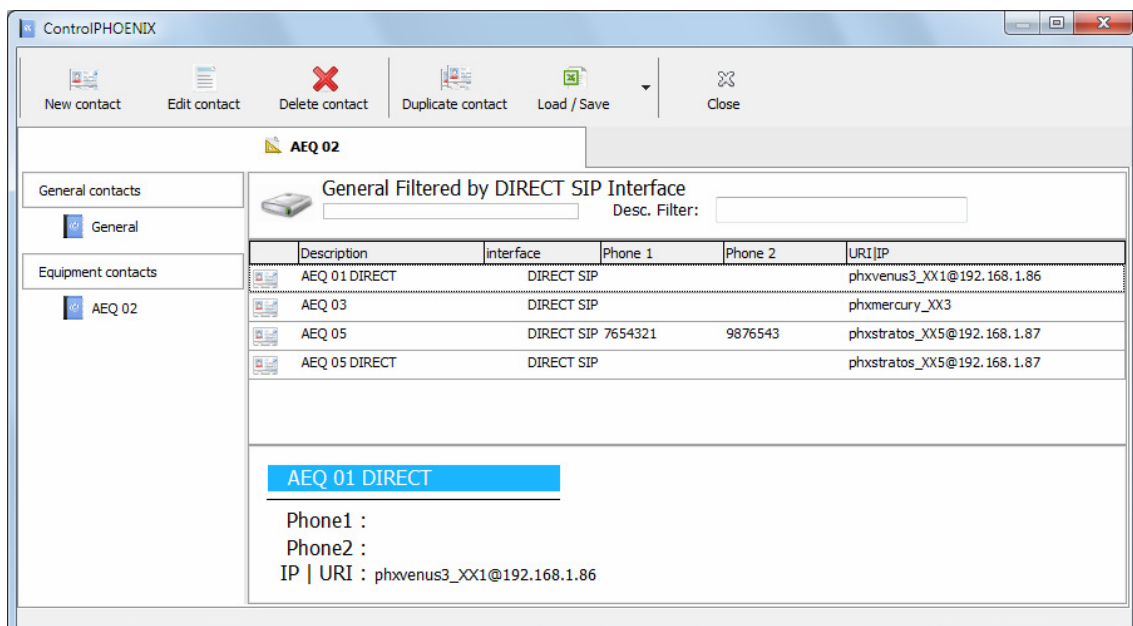


By clicking on that button, the call window shows up, and the **"Call to:"** field can be edited with keyboard. This field must be filled in with a URI or telephone number (depending on the type of selected communications interface).

The URI or telephone number can also be directly imported from the call book (by clicking on the  button).

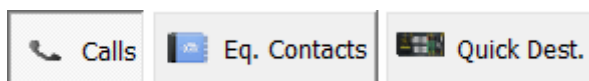
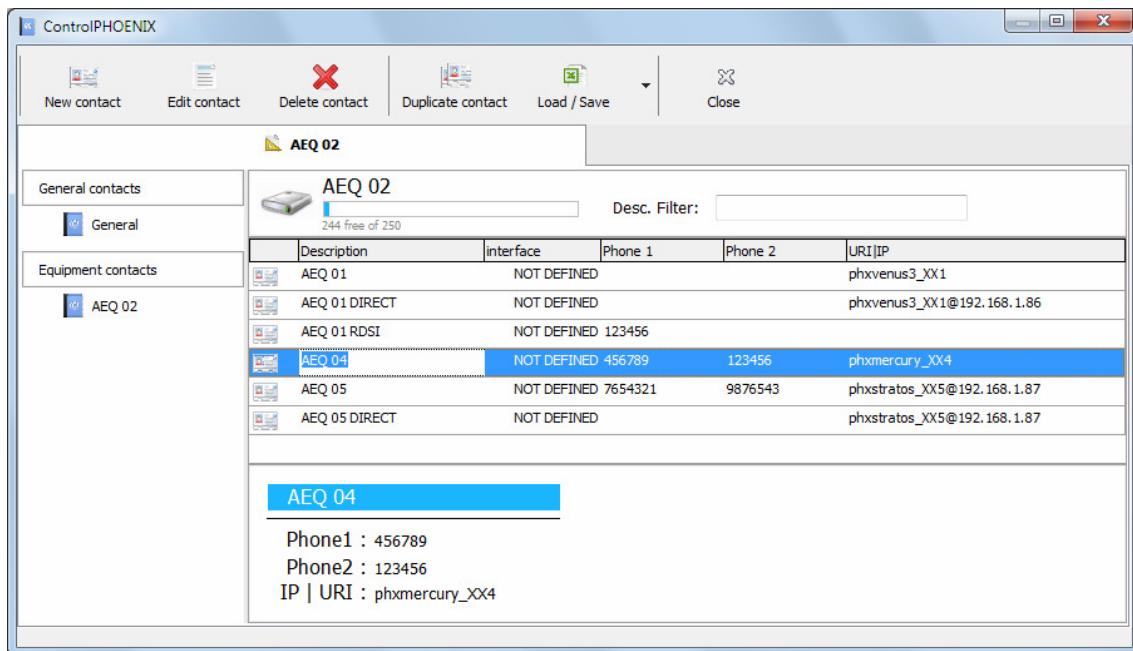
The imported number or URI will appear in the **"Call to:"** field, and then the call can be established by clicking on **"Call"** green button.

 By clicking on this button, all the contacts of the **general agenda** will appear filtered depending on the interface type configured in the calling channel at that moment, provided that those contacts have been created defining the "Interface" field (and its associated advanced parameters). Besides, the contacts created without defining the "Interface" field will also be shown, but regardless of the compatibility with the actual channel and intended communication type.



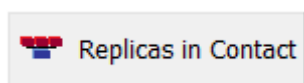
When one of these **advanced contacts** is selected, the configured interface of the unit will not change but the following parameters will be configured: the encoding mode/profile and, if defined, the replicas configuration (only in RTP mode) and the SIP account (only in Proxy SIP mode; in that case, the unit will try to register and, after doing it, the call will be launched). When the configured interface of the unit is not compatible with the one configured in the contact, this will not be shown in the previous list; when the encoding mode or profile defined in the contact is not available in the calling unit, the contact will be shown in the list but, if selected, the call will not be launched.

From this window you can also access to the **individual agenda** of the selected equipment, showing then all the defined contacts, regardless of the compatibility with the actual channel and intended communication type.



By clicking on these buttons, all the numbers and URI corresponding to last calls ("Calls"), the ones from the individual call book ("Eq. Contacts") or the IP units

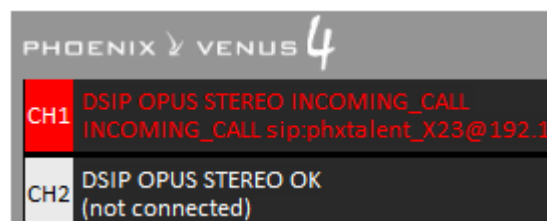
that can be reached ("Quick Dest.") will appear respectively, but only those that are compatible with the actual channel and intended communication type.



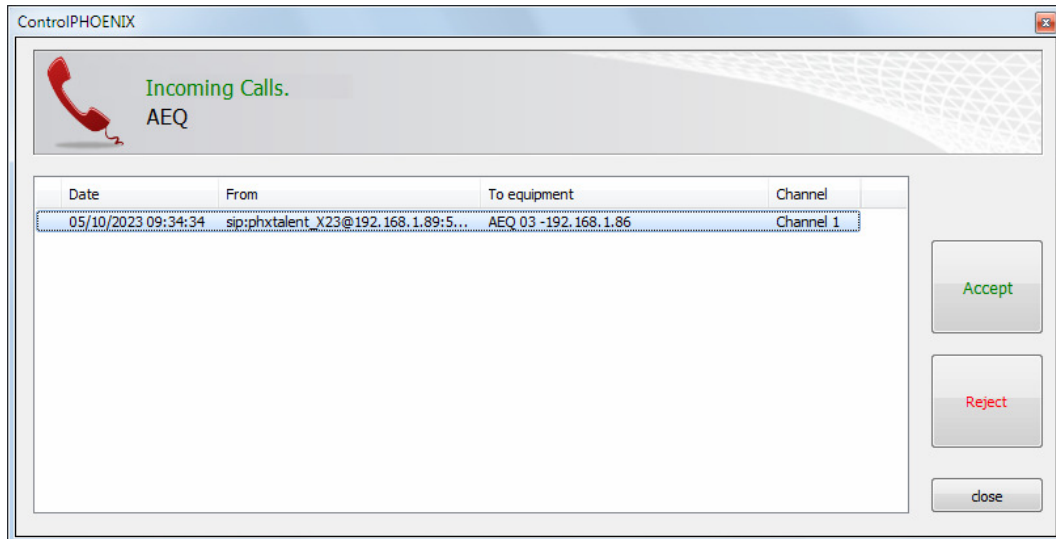
When the configured interface of the calling channel is RTP a fourth button will also appear in order to check the replicas associated to the contact (when defined) but only when it's selected from the

general agenda (by means of the  button).

Once the destination is selected, when you press the green **"Call"** button, the receiving codec will show the **"INCOMING CALL"** message in its corresponding channel control window, as well as the call origin (in the codec that calls the **"REMOTE RINGING"** message and the call destination will be displayed).

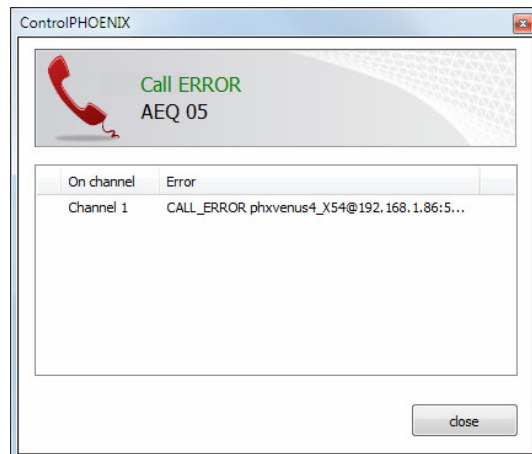


In addition, an incoming call window will appear and the application buzzer will play in order to alert the user. Besides the equipment's buzzer will sound (if activated, see section 6.4.5).



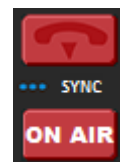
You can accept the incoming call either by pressing the **"Accept"** button in that window or by pressing the CALL button of the equipment that's receiving the call.

When the call is rejected (by means of the **"Reject"** button), a window will appear in order to inform about the call error:

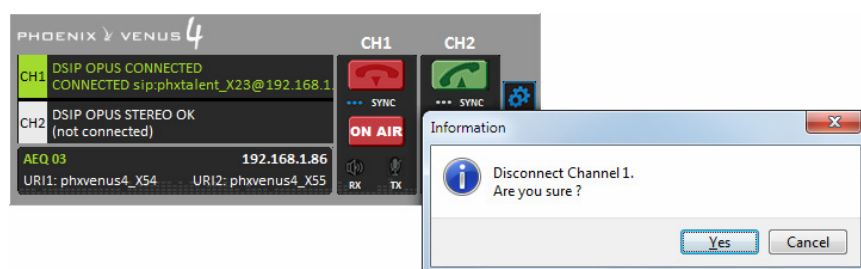


When that window is closed (by means of **"close"** button) and there are still unanswered calls, the "Incoming calls" icon of the upper menu (see section 5.1.6) changes its aspect from shaded grey to a variable color in order to signal this situation (by pressing it the incoming calls list window appears again).

On the other hand, when the call is accepted, the **"SYNC"** LED should turn blue on both codecs (calling and called). Now **"ON AIR"** can be activated to route the incoming audio from the other codec to the outputs. Note that **"ON AIR"** button also enables the audio transmission from the input to the other codec through the communications channel. In the case of **Phoenix ALIO** and **TALENT**, without **"ON AIR"** buttons, that activation and routing is done through the mixer control window (for more information, please consult the user's manual).

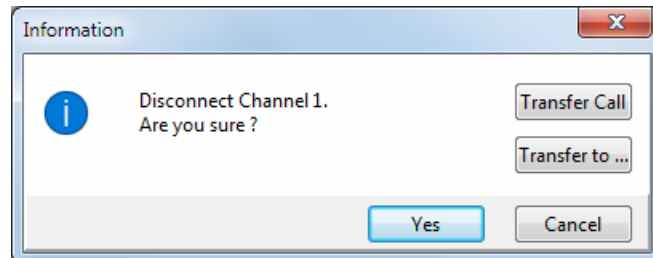


In order to hang up the call, you simply need to click on the CALL button corresponding to the connected channel again, and the call will be hang up once confirmation is provided.



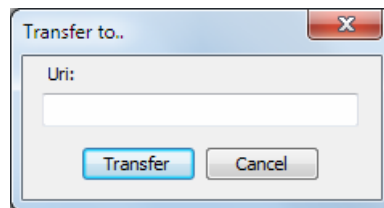
6.7.1. Call forwarding in PROXY SIP.

Version **2.2.2.4** (or higher) of **AEQ ControlPHOENIX** application, when acting on Phoenix audiocodecs with CPU version **10.60** or higher, allows call forwarding in PROXY SIP, forwarding the call established at a given time in an audiocodec towards a PHONE. The procedure is as follows: when a call is established in PROXY SIP mode and you try to hang up that call, the following confirmation message appears:

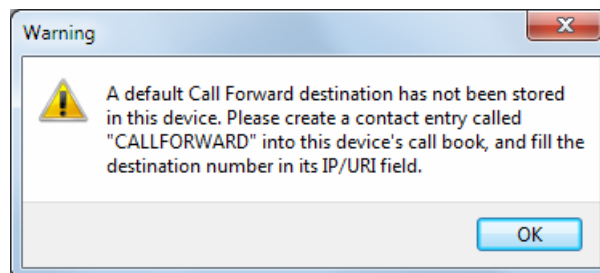


As previously indicated, in that window you can confirm to hang up the call by pressing "Yes" or cancel that process by pressing "Cancel". The latest firmware and software versions show also the "**Transfer Call**" and "**Transfer to...**" options.

By pressing "**Transfer to...**" the following window appears that allows you to enter the telephone number you want to transfer the call and confirm that process by pressing the "Transfer" button (the current call is then cut and transferred to the indicated phone). By pressing "Cancel" the process is aborted and the hang up/transfer window is closed.



By pressing "**Transfer Call**" the call is automatically transferred to the telephone number defined in the "**CALLFORWARD**" contact of the unit (see section 5.1.7.1.1 of this manual). When this contact is not present in the unit, the following warning appears so we can create it (by pressing "OK" this window and the hang up/transfer window are closed):



7. SETTING UP PHOENIX STRATOS, PHOENIX STUDIO AND PHOENIX VENUS V4+ UNITS TO USE REMOTE CONTROL.

First make sure that the firmware of the unit is upgraded to the last available versions.
IMPORTANT NOTE: Please, don't upgrade the equipment firmware without consulting AEQ Technical Service previously.

Remote control configuration can be performed directly from the application (once it has managed to connect to the equipment, in order to adjust the connection mode) or from the Phoenix Stratos, Phoenix Studio and Phoenix Venus V4+ front panel itself.

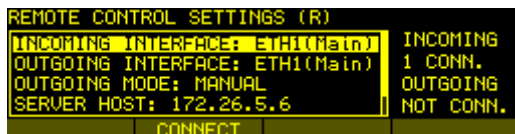
7.1. Configuration from the equipment's front panel (Phoenix Stratos, Phoenix Studio and Phoenix Venus V4+ only).



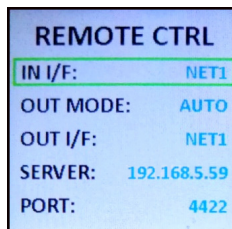
Configuration of the remote control can be found in submenu MAINTENANCE under SYSTEM menu.



In the case of Phoenix V4+ it can be found in submenu DEVICE CONFIG under CONFIG menu.



The REMOTE CONTROL SETTINGS submenu allows the user to configure how the connection between equipment and PC is established.



In the case of Phoenix V4+ the REMOTE CTRL submenu looks like this.

There are two types of remote control connections: incoming (PC to audiocodec) and outgoing (audiocodec to PC). The equipment can be configured to accept incoming connections from both Ethernet ports (only in Phoenix Studio and Phoenix Venus V4+), from only one or to reject any incoming connection ("NONE"). Up to three incoming connections can be established.

Additionally, the unit can be set up to with one (and only one) PC application (that we will refer to as "remote control server"). In order to do that, we need to configure the name or IP of the server as well as the control port (4422 by default), as well as the Ethernet port that we want to use to control the equipment. There are two connection modes: manual and automatic. If the automatic mode is applied, this results in that the equipment continuously will try to connect (whenever not connected).



The right part of the screen (only in Phoenix Stratos and Phoenix Studio) shows the number of incoming connections as well as the status of the outgoing connection (not connected, trying to connect, connected...).



Besides, all the screens of Phoenix Stratos or Phoenix Studio will show a "R" symbol in the title bar when it is remotely controlled by at least one application, so the user is aware that several people can be operating the unit at the same time simultaneously.

7.2. Configuration from the remote control software.

All Phoenix units are configured from Factory with a default IP (check its user manual for more details) so the remote control connection is immediate by only configuring the PC network interface so it falls within the same network (so it can successfully issue a 'ping' command to the audiocodec).

Once the connection is established, adjustments can be made to the way the unit connects to the software, etc. without having a front panel by means of the **"Network"** → **"Remote control"** menu, as explained in section 6.5.3 of this manual.

8. A.E.Q. GUARANTEE.

AEQ warrants that this product has been designed and manufactured under a Quality Assurance System. AEQ therefore warrants that the necessary test protocols to assure the proper operation and the specified technical characteristics of the product have been followed and accomplished.

This includes that the general protocols for design and production and the particular ones for this product are conveniently documented.

1.- The present guarantee does not exclude or limit in any way any legally recognized right of the client.

2.- The period of guarantee is defined to be twelve natural months starting from the date of purchase of the product by the first client. To be able to apply to the established in this guarantee, it is compulsory condition to inform the authorized distributor or –to its effect- an AEQ Sales office or the Technical Service of AEQ within thirty days of the appearance of the defect and within the period of guarantee, as well as to facilitate a copy of the purchase invoice and serial number of the product.

It will be equally necessary the previous and expressed conformity from the AEQ Technical Service for the shipment to AEQ of products for their repair or substitution in application of the present guarantee.

As a consequence, returns of equipment that does not comply with these conditions will not be accepted.

3.- AEQ will at its own cost repair the faulty product once returned, including the necessary labour to carry out such repair, whenever the failure is caused by defects of the materials, design or workmanship. The repair will be carried out in any of the AEQ authorized Technical Service Centres. This guarantee does not include the freight charges of the product to or from such Authorized Technical Service Centre.

4.- No Extension of the Guarantee Period for repaired product shall be applied. Nor shall a Substituted Products in application of this Guarantee be subject to Guarantee Period Extension.

5.- The present guarantee will not be applicable in the following situations:

Improper use or Contrary use of the product as per the User or Instruction Manual; violent manipulation; exhibition to humidity or extreme thermal or environmental conditions or sudden changes of such conditions; electrical discharges or lightning; oxidation; modifications or not authorized connections; repairs or non-authorized disassembly of the product; spill of liquids or chemical products.

6.- Under no circumstances, whether based upon this Limited Guarantee or otherwise, shall AEQ, S.A. be liable for incidental, special, or consequential damages derived from the use or from the impossibility of using the product.

AEQ shall not be liable for loss of information in the disks or data support that have been altered or found to be inexact, neither for any accidental damage caused by the user or other persons manipulating the product.