



**Ultra-Portable IP Audiocodect that is easy to configure and use
Optimized for OPUS encoding algorithms**

**USER'S MANUAL
ED. 03/23**

V. 1.0 - 01/03/2023

Firmware Versions:	CPU 10.20 / DSP 3.36 or higher
Software Version:	AEQ ControlIPHOENIX 2.2.2.0 or higher
App Version:	AEQ TALENT PILOT 1.0.2 or higher

CONTENTS

1. INTRODUCTION.....	5
1.1. General description.....	5
1.2. Technical characteristics.....	5
1.3. Available encoding algorithms.....	6
1.4. Audio Routing.....	7
1.5. Compatibility with other AEQ codecs.....	7
1.6. Compatibility with third-party codecs.....	8
2. PHYSICAL DESCRIPTION OF THE UNIT.....	9
2.1. Description of the right panel and connections.....	9
2.1.1. Headphone output.....	9
2.1.2. Microphone input.....	9
2.2. Description of the back panel and connections.....	10
2.2.1. Line input.....	10
2.2.2. Line output.....	10
2.2.3. "RESET" pushbutton.....	10
2.2.4. Ethernet port.....	11
2.2.5. Micro USB connector.....	11
2.2.6. Power supply switch.....	11
2.2.7. Co-Axial power supply Jack connector.....	11
3. USER INTERFACE DESCRIPTION. MANUAL CONTROL.....	12
3.1. Power ON indicator.....	12
3.2. Level indicator.....	12
3.3. Established link indicator.....	12
3.4. "HELP" Key.....	13
3.5. Microphone and Panorama encoder.....	13
3.6. Headphone and output line / Bluetooth Encoder.....	13
3.7. Call control keys.....	14
3.8. Microphone ON / OFF (MUTE) key.....	14
3.9. Status indicators.....	14
4. CONFIGURATION AND OPERATION OPTIONS FROM "AEQ ControlPHOENIX" REMOTE CONTROL SOFTWARE.....	15
4.1. Individual codec control window.....	15
4.1.1. CONFIGURATION Menu.....	17
4.1.2. Mixer control window.....	18
4.1.3. Vumeters window.....	19
4.2. Connection modes.....	20
4.2.1. RTP Point to Point (RAW).....	20
4.2.2. PROXY SIP.....	21
4.2.3. DIRECT SIP.....	23
4.2.4. Sending audio to multiple destinations: Broadcast, Multicast and Multi-unicast.....	24
4.3. NAT TRAVERSAL.....	28
4.3.1. Operation without NAT: "OFF (there is no NAT)".....	28
4.3.2. Manual NAT: "MANUAL (router configuration)".....	28
4.3.3. AUTO 1 (local network audio).....	30
4.3.4. AUTO 2 (local network audio).....	30
4.3.5. AUTO 3 (audio over internet).....	30
4.3.6. AUTO 4 (audio over internet).....	30
4.4. FEC modes and reception buffer configuration.....	30
4.6. Ethernet Port configuration.....	33
4.7. SNMP configuration.....	33
4.8. Password configuration.....	35
4.9. User interface lock.....	35

5. CONFIGURATION AND OPERATION OPTIONS FROM "AEQ TALENT PILOT" REMOTE CONTROL APPLICATION.	38
5.1. Downloading "AEQ TALENT PILOT".	38
5.2. Running "AEQ TALENT PILOT" for the first time.	38
5.3. Main Screen.	39
5.3.1. Upper area.	40
5.3.2. Main area.	40
5.3.3. Lower area.	42
5.4. Call menu.	42
5.5. Audio adjustments.	45
5.5.1. Inputs adjustment menu.	45
5.5.2. Outputs adjustment menu.	46
5.6. Settings screen.	47
6. QUICK START GUIDE.	51
6.1. First steps. "AEQ TALENT PILOT" control App.	51
6.2. Connecting to a network.	52
6.3. "AEQ ControlPHOENIX" configuration software.	52
6.4. Making your first test call.	52
6.5. Pairing for audio exchange.	53
7. CONTROL TERMINAL OVER WEB BROWSER.	55
7.1. Upgrading system firmware.	55
7.2. Configuring the IP parameters and the MAC address associated with the Ethernet interface.	56
7.3. Technical Service and on-line manuals.	57
7.4. Saving and loading configurations.	57
7.5. Status menu.	58
7.6. SNMP.	58
7.7. Remotely rebooting the equipment.	59
8. TECHNICAL SPECIFICATIONS	60
9. A.E.Q. WARRANTY.	61
APPENDIX A: GENERAL CHARACTERISTICS OF ENCODING MODES.	62
APPENDIX B: PROTOCOLS ASSOCIATED WITH IP COMMUNICATIONS.	64
B1. Circuit switching versus packet switching.	64
B1.1. Circuit switching.	64
B1.1.1. Advantages:	64
B1.1.2. Disadvantages:	65
B1.2. Packet switching.	65
B1.2.1. Switching modes.	65
B1.2.2. Advantages:	65
B1.2.3. Disadvantages:	66
B2. IP protocol.	66
B2.1. IP addressing.	67
B2.2. Unicast vs. Multicast.	67
B3. RTP protocol.	68
B3.1. Default TALENT configuration.	69
B4. SIP protocol.	69
B4.1. Working modes.	69
B4.2. Possible work scenarios.	71
B4.3. TALENT default SIP configuration.	71
B5. STUN protocol.	72
APPENDIX C: PORTS USED BY TALENT.	74
APPENDIX D: APPLICATION NOTES GUIDE.	75
D1. Internet connection using standard cable access.	75

D2. Special applications using different kinds of Internet physical accesses, or point to point connections.....	75
APPENDIX E: TALENT TROUBLESHOOTING GUIDE.....	76
APPENDIX F: ADDITIONAL INFORMATION.....	81

1. INTRODUCTION.

1.1. General description.

AEQ TALENT is a single-channel audio codec with professional microphone input and headphone output. Also, it counts with auxiliary 3.5mm jack line input and output and Bluetooth connectivity.

AEQ TALENT is optimized for OPUS encoding algorithms, but it is also compatible with other AEQ and third-party audiocoders, as it also features AEQ LD Extend modes and the mandatory algorithms according to EBU-TECH 3326 specification from EBU N/ACIP work group.

When connecting to another AEQ codec, users can take advantage of an exclusive set of tools that makes initiating communications and control of the unit a simple task:

- The "**SmartRTP**" proprietary call-initiation protocol that simplifies connection to compatible codecs.
- "**AEQ ControlPHOENIX**", remote control Software that allows for the remote operation and adjustment of the unit from your station. ControlPhoenix allows you to control everything related to the call initiation process and also the adjustment of all audio parameters and the local audio routing of TALENT.
- It is also possible to control the unit using "**AEQ TALENT PILOT**" **App** for Android and iOS, running on a Bluetooth-linked smart device. It is also possible to transfer audio between a smart device and TALENT audiocoder.
- "**HELP**" function, that allows the journalist to use the system to request for assistance from the station when facing an unexpected situation.

It is recommended to carefully read this manual and the "AEQ ControlPHOENIX" user's guide before using the unit, although chapter 6 offers a quickstart user's guide.

1.2. Technical characteristics.

- **Microphone input:**
 - Female XLR-3 connector.
 - Input impedance: 2 K Ω .
 - High quality, low noise, electronically balanced preamplifier
 - Valid for most microphones with nominal 48V Phantom.
 - Gain between 0 and +60dB, adjustable from the front panel, mobile App or control software.
 - MUTE key.
- **Stereo ground-referenced line input:**
 - 3.5mm female jack connector.
 - Input impedance >15 K Ω .
 - Input level (LINE IN Gain set to 0 dB):
 - Nominal: 178 mV RMS (-15 dBV).
 - Maximum: 1V RMS (0 dBV).
 - Gain selectable between -6dB and +20dB from mobile App or control software. Line & Bluetooth gain selection is common. These inputs are mixed together
 - Compatible with PCs, tablets, phones and other sound players.
- **Bluetooth 5.0 audio input. Stereo A2DP:**
 - Gain selectable between -6dB and +20dB from control software. Line & Bluetooth gain selection is common.
 - A2DP profile for high-quality audio reception (Bluetooth 5.0 with AAC encoding, among others).
 - HFP voice profile (with G711 or G722 encoding, depending on the phone or conferencing App used).
 - This input is mixed with audio coming from the line input.

- **Stereo Headphone output:**
 - 1/4" stereo female jack connector.
 - Power ≥ 100 mW (50 mW max when using 16 Ω headphones).
 - Compatible with low or high impedance headphones between 16 and 500 Ω .
 - Volume control and TX/RX panorama selection from the front panel, by App and by software.
- **Stereo ground-referenced line output:**
 - 3.5mm female jack connector.
 - Output impedance: $< 50 \Omega$.
 - Adjustable gain from front panel, mobile app and control software. There is a common input level adjustment for line input and Bluetooth incoming audio.
 - Maximum output level: 835mV RMS (-1.5dBV) for THD $< 0.5\%$.
 - Compatible with PCs, tablets, phones and other sound recorders.
 - This output follows the audio signal sent to the outgoing Bluetooth connection.
- **Bluetooth 5.0 audio output. Stereo A2DP:**
 - Adjustable gain from front panel, mobile app and control software. There is a common input level adjustment for line output and Bluetooth outgoing audio.
 - A2DP profile for high-quality audio reception (Bluetooth 5.0 with AAC encoding, among others).
 - HFP voice profile (with G711 or G722 encoding, depending on the phone or conferencing App used).

Communications interface:

- IP interface: 10/100 Mbps Ethernet port, RJ45 connector. The **default IP address** for TALENT is **192.168.1.89**, although all devices will leave the factory with **DHCP** option activated (configured to obtain its complete IP configuration from a DHCP server). This interface provides access to:
 - Wired networks: Local area networks, Cable ADSL, Fiber and VLANs.
 - Satellite: an external satellite interface can be connected to the IP interface.
 - 3G / 4G / 5G telephony: a 3G, 4G or 5G modem can be connected to the IP interface.
 - Wireless data links: a "wireless bridge", WiMax or WiFi antenna can be connected to the IP interface.

Other Features:

- Local interface with keys and encoders.
- Three-level LED vumeter at the front panel for program TX level indication.
- Received audio signal presence indicator.
- Operating temperature: -10 to +45 °C (14 to 114 °F).
- Dimensions: Width x Depth x Height: 130x195x45mm.
- Weight: 620gr.
- Power supply: 5 to 15 VDC for external adapter (90-263V 50/60 Hz input) or 5V through USB interface. Power consumption: 3.5 W max.

AEQ offers the "SmartRTP" call initiation protocol in order to greatly simplify the operation of the audiocodec.

AEQ also offers two SIP servers free of charge as a standard service for the users of Phoenix ALIO. One of them is configured as main and the other is provided as a backup. More information can be found in Appendix B.

1.3. Available encoding algorithms.

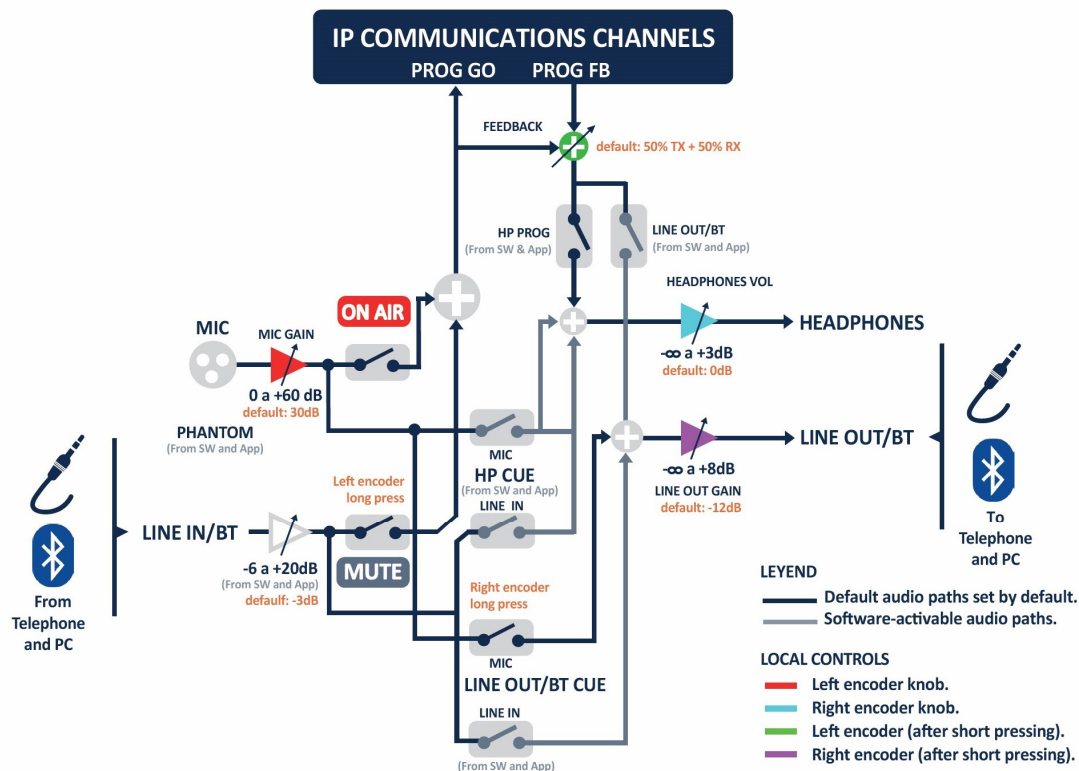
- **OPUS** with $F_s = 48$ KHz, mono, stereo, with 4 mono and 3 stereo presets. Bit rates between 12 and 192 Kbps, very low delay and audio bandwidth between 6 and 20 KHz.
- **G711** A-law, u-law (64 Kbps, low delay, 3.5 KHz audio bandwidth).

- **G722** (64 Kbps, low delay, 7 KHz audio bandwidth).
- **AEQ-LD** with $F_s=16, 32$ or 48 KHz, mono or stereo. Available bit rates between 64 and 384 Kbps, audio bandwidth between 7 and 19 KHz.
- **MPEG 1 and 2 - LII**, with F_s between 16 and 48 KHz, mono, stereo, dual channel and Joint stereo. Binary bit rate between 64 and 384 Kbps. Audio bandwidth between 10.5 and 16.5 KHz.
- **PCM (linear)**: very low delay, transparent quality. $F_s=48$ or 32 KHz with 12, 16, 20 or 24 bits/sample, mono or stereo (bit rates between 576 and 2304 Kbps), audio bandwidth between 16 and 20 KHz.

The implementation of other coding algorithms may be evaluated according to each customer's specific requirements. Please ask AEQ Sales Department or authorized dealers in order to get more information.

1.4. Audio Routing.

The following diagram shows the device's audio capabilities, as well as how to access each adjustment from the front panel and factory default audio settings.



1.5. Compatibility with other AEQ codecs.

TALENT offers the possibility to connect to other AEQ codecs.

It is compatible with Phoenix ALIO, MERCURY, STUDIO, VENUS, VENUS V2, VENUS V3, VENUS V4, VENUS V4+, STRATOS and MOBILE.

The "**SmartRTP**" mode and **OPUS** encoding algorithm can be used with Phoenix ALIO, MERCURY, STUDIO, VENUS, VENUS V2, VENUS V3, VENUS V4, VENUS V4+ and STRATOS updated to latest firmware versions.

1.6. Compatibility with third-party codecs.

TALENT is a portable IP audiocodec compatible with EBU-TECH 3326 technical specification from EBU N/ACIP workgroup. This technical specification was developed to guarantee compatibility between equipment from different manufacturers in applications for professional-quality audio contribution over IP networks. Therefore it is possible to connect TALENT with any codec from other manufacturer over IP provided that this unit has been developed according to N/ACIP (please check third-party codecs technical specifications).

2. PHYSICAL DESCRIPTION OF THE UNIT.

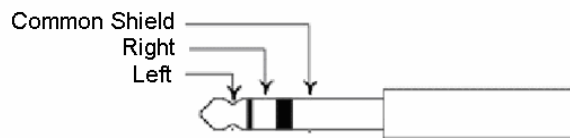
First of all, it is necessary to become familiar with the connectors and other elements present in the back and right panels of the unit, in order to understand the wiring and installation required to use TALENT.

2.1. Description of the right panel and connections.



2.1.1. Headphone output. **A**

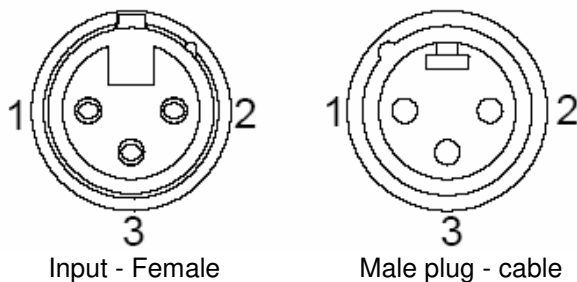
1/4" Headphone Jack. Unbalanced connection.



1/4" Jack pinout

2.1.2. Microphone input. **B**

XLR-3 female connector. Balanced connection.



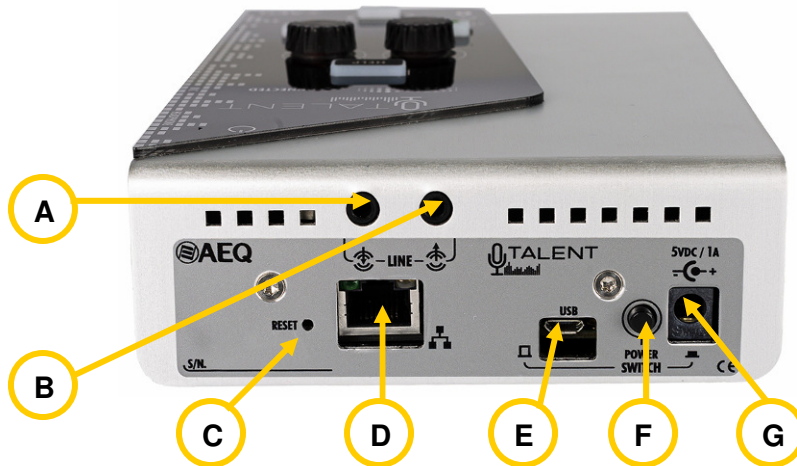
(Connector as seen by the soldered side)

XLR-3 pinout

- Pin 1 → Ground
- Pin 2 → + Input
- Pin 3 → - Input

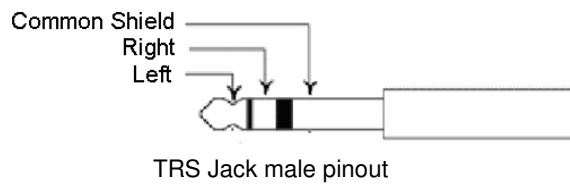
The microphone input features a high quality low-noise preamplifier and is able to provide Phantom supply, software switchable, in order to offer compatibility with both dynamic or condenser microphones (valid for most microphones with nominal 48V Phantom). The preamplifier gain range is from 0 to 60 dB, making it suitable for a large range of microphone models available in the market.

2.2. Description of the back panel and connections.



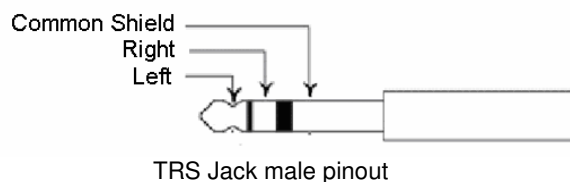
2.2.1. Line input. **A**

Jack TRS 3.5 mm female connector. Unbalanced connection.



2.2.2. Line output. **B**

Jack TRS 3.5 mm female connector. Unbalanced connection.



2.2.3. "RESET" pushbutton. **C**

This push-button allows the user to **reset** the device configuration in several ways, according to this sequence (for this there must be no call established on the device):

- 1) Hold it pressed until a "beep" is emitted. If the push-button is released at that moment, the unit will return a different confirmation tone and will return to **DHCP** mode, without any further changes.
- 2) If the push-button is held pressed for a longer time until it emits 2 beeps, and it is released at that moment, the unit will return a different double confirmation tone and will reboot adopting the following default Ethernet configuration: **192.168.1.89 fixed IP address** (DHCP mode is turned off), **/24 network mask (255.255.255.0)**, **192.168.1.1 gateway** and **8.8.8.8 DNS server**, without further alteration.
- 3) If, the push-button is held until the unit beeps three times and then released, after a different triple confirmation tone it will perform a full **FACTORY CONFIGURATION RESET**, although **ONLY** the following parameters will remain unchanged:

- IP configuration.
- Incoming control accepted ports and default gateway.
- Outgoing control parameters (activation, IP and port).
- TCP or UDP protocol selection.

IMPORTANT NOTICE: As part of this configuration, **SIP users** established from factory for the unit are lost when activating this last option, so it is important to take note of their values in order to be able to re-configure them later on.

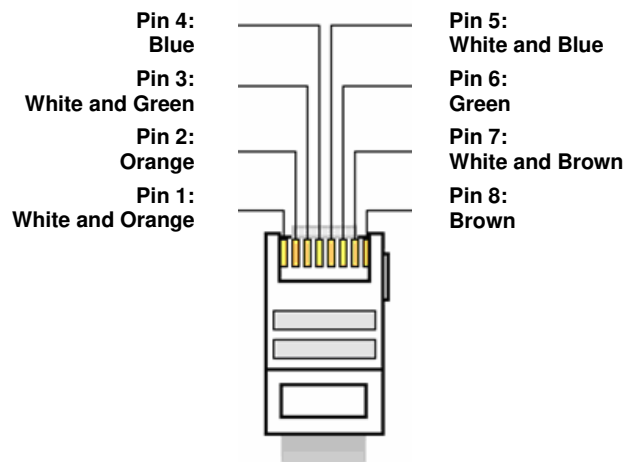


2.2.4. Ethernet port. **D**

TALENT features one Ethernet port. Using this port, the unit can be connected to a LAN or WAN network and send / receive audio over IP. This port is also used to configure and administrate the unit from one or more computers using the remote control software. The **default IP address** for TALENT is **192.168.1.89**, although all devices will leave the factory with **DHCP** option activated (configured to obtain its complete IP configuration from a DHCP server).

Please refer to "AEQ ControlPHOENIX" application manual.

The connector is a standard RJ45 10/100 BT, type, with the following pin assignment:



RJ45 connector pinout

2.2.5. Micro USB connector. **E**

This connector is reserved for factory maintenance uses and for external power supply using a power-bank, AC/DC adapter/charger or computer, provided that 5V and a minimum current of 1A are provided.

2.2.6. Power supply switch. **F**

Allows for the selection of the power source (DC co-axial jack or micro USB port). Under normal circumstances, only one of the plugs will be connected, so it will act as a simple power on switch.

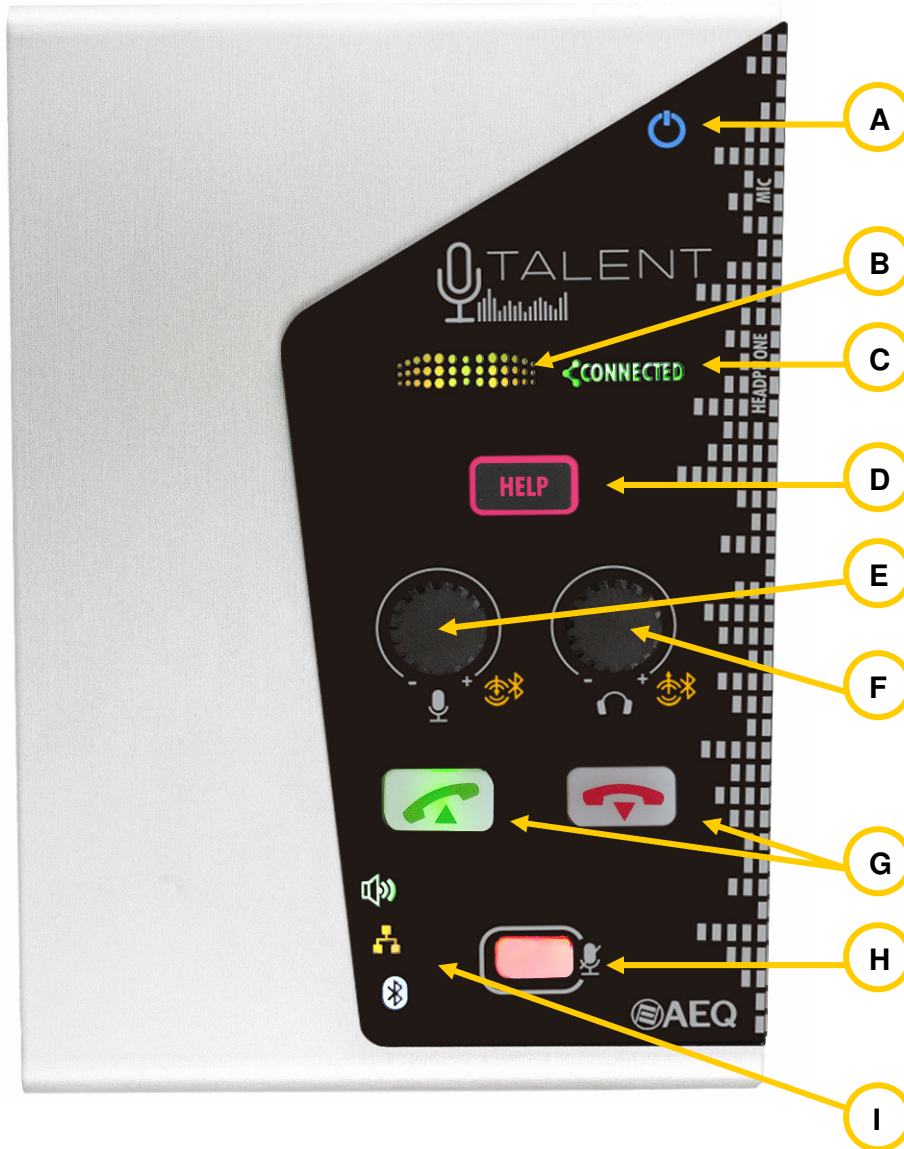
If the power supply is provided by means of the DC co-axial jack, the unit turns on when the push-button is pressed, while, on the other hand, if power supply is provided through the USB connector, the unit will turn on when the push button is in its outer position.

2.2.7. Co-Axial power supply Jack connector. **G**

5mm/2.1mm co-axial jack for external power supply. Voltage range: 5 to 15VDC, 1A max (@5Vdc). Central pin is positive.

3. USER INTERFACE DESCRIPTION. MANUAL CONTROL.

Configuration and operation of the **TALENT** unit can be done **locally** either using the controls and indicators featured in the front panel, **locally** using "**AEQ TALENT PILOT**" App or **remotely**, using the "**AEQ ControlPHOENIX**" application (Control and Configuration software for AEQ TALENT, Phoenix STRATOS, STUDIO, MERCURY, VENUS, VENUS V2, VENUS V3, VENUS V4, VENUS V4+ and ALIO audiocoders). This chapter describes the first option.



3.1. Power ON indicator. **A**

3.2. Level indicator. **B**

It operates as a simple vumeter if any of the inputs is ON AIR. Presence of signal is first displayed in green when it is above -58 dBFS, changing to green-yellow when it is above -22 dBFS and adding red when it is above -12 dBFS, indicating danger of over-modulation.

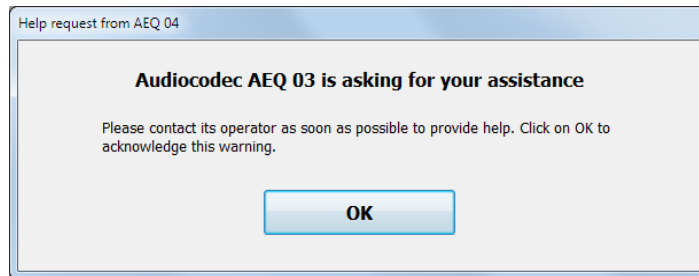
3.3. Established link indicator. **C**

3.4. "HELP" Key.

D

This key sends a notification to the remote control software provided that this is in use. This way, a 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, 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 TALENT user.



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

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

On the other hand, a long-pressing of the "HELP" button alternatively turns on or off the Bluetooth module integrated into the unit.

3.5. Microphone and Panorama encoder.

E

If this encoder is turned, the microphone input gain is controlled.

A brief pressing on the encoder push-button will illuminate the associated **yellow** LED, indicating that, when turning the encoder, adjustment now acts on the TX/RX panorama for the outputs currently listening to the PROGRAM bus. 3 seconds after the last adjustment of the knob the yellow LED will extinguish and the encoder returns to normal (microphone gain) adjustment mode.



Pressing the encoder push-button for a while acts on the line / Bluetooth input ON AIR setting, and, when ON, the associated LED illuminates **red**. Pressing it again for a while turns off the LED and stops sending this input to program. In both situations, the encoder knob rotation still works as described above.



Line input / Bluetooth level adjustment can be performed from "AEQ ControlPHOENIX" control software.

3.6. Headphone and output line / Bluetooth Encoder.

F

The headphone output level is adjusted when turning this encoder knob.

A brief pressing on the encoder push-button will illuminate the associated **yellow** LED, indicating that, when turning the encoder, adjustment now acts on line / Bluetooth output level. 3 seconds after the last adjustment of the knob the yellow LED will extinguish and the encoder returns to normal (headphone volume) adjustment mode.



Pressing the encoder push-button for a while acts on the activation of the local microphone sending to the line / Bluetooth output, and, when ON, the associated LED illuminates **red**. In both situations, the encoder knob rotation still works as described above.



Full control of the signal/s sent towards headphone output or line / Bluetooth output is provided from "AEQ ControlPHOENIX" software or "AEQ TALENT PILOT" App. It is always possible to activate / deactivate MIC send to line / Bluetooth output, however, as explained above, with long pressings of this encoder push-button.

3.7. Call control keys.

The leftmost key (green telephone handset) allows the user to call to the latest configured destination by pressing it for a while. This key also allows for acceptance of incoming calls by shortly pressing it. The key will blink green while the call is being established or ringing, and stays illuminated when it is successfully established.

The rightmost key (red telephone handset) allows the user to hang up an already established call by means of a long-press. On the other hand, this key also rejects an incoming call, also with a long-pressing. This key blinks in red during the release of the call, and it finally extinguishes.

3.8. Microphone ON / OFF (MUTE) key.

This key puts or removes the microphone input ON AIR. The key illuminates in red when the microphone is sent to program.

3.9. Status indicators.

Three general indicators can be seen in this area:



Incoming audio presence indicator (Rx). It is possible to adjust its threshold level and expiration time using the remote control software (By default it illuminates when the received signal is above -57 dBFS and extinguishes when it is below that level for longer than 2 seconds.).



IP ready indicator. This indicator is off when there is no LAN connection, and stays illuminated when the unit is ready to establish a communication. When it is configured in **Proxy SIP** mode, this implies that there is Internet connectivity and the device has successfully registered in a SIP server. If it blinks, that means that registration failed, but the unit at least has Ethernet link. On the other hand, if the unit is operating in **RTP Raw** mode (with DHCP option active), when this indicator blinks that means that there is physical LAN link but the unit hasn't been able to get proper IP configuration yet.



Bluetooth indicator. This indicator is off when the integrated Bluetooth module is off. It blinks blue when it is on and ready to pair for audio, and stays illuminated blue when the unit has been successfully paired for audio with a smart device (usually a mobile Phone). The red color is associated to the control link: so if the unit is paired both for audio and control (TALENT Pilot App is connected), this indicator will look steady pink, while if it is only paired for control it will blink between red and pink.

When activating Bluetooth, two visible devices should appear. The first one ("AEQ_TALENT_xxxx", where xxxx is the device's serial number) allows for Bluetooth audio connectivity after pairing it. The second one ("AEQ_TALENT_BLE_xxxx") allows for control from "AEQ TALENT PILOT" App (in this case, pairing will be automatically performed when opening the App and selecting this device).



AEQ_TALENT_xxxx



AEQ_TALENT_BLE_xxxx

4. CONFIGURATION AND OPERATION OPTIONS FROM "AEQ ControlPHOENIX" REMOTE CONTROL SOFTWARE.

Configuration and operation of TALENT can be carried out remotely by means of "AEQ ControlPHOENIX" software (AEQ TALENT, Phoenix ALIO, STRATOS, STUDIO, MERCURY, VENUS, VENUS V2, VENUS V3, VENUS V4 and VENUS V4+ Audiocodex Configuration and Control Software).

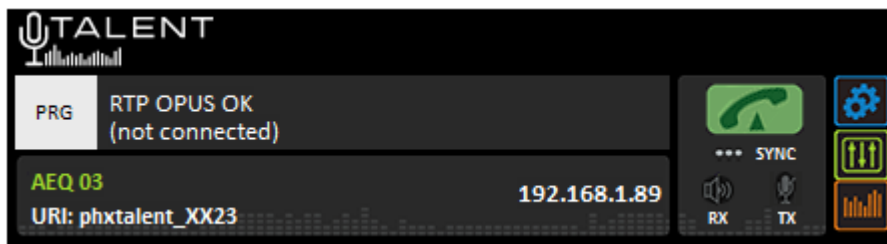
The version that is provided together with the equipment (**2.2.2.0** or higher) can control up to **2 units** per software instance. If you need to manage more than 2 audiocodex at the same time, please contact AEQ sales department to purchase a multicodec-license for "AEQ ControlPHOENIX".

Furthermore, from version **2.2.1.50** of the application is **NOT** possible to control units with CPU 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.

Please have the user manual of this application at hand. Install and configure it and add the equipment to the controlled equipment list in order to follow the explanations provided in this and following chapters step by step. This manual will describe particular TALENT options only, as well as some important procedures, while the detailed operation is detailed in "AEQ ControlPHOENIX" user's manual.

4.1. Individual codec control window.

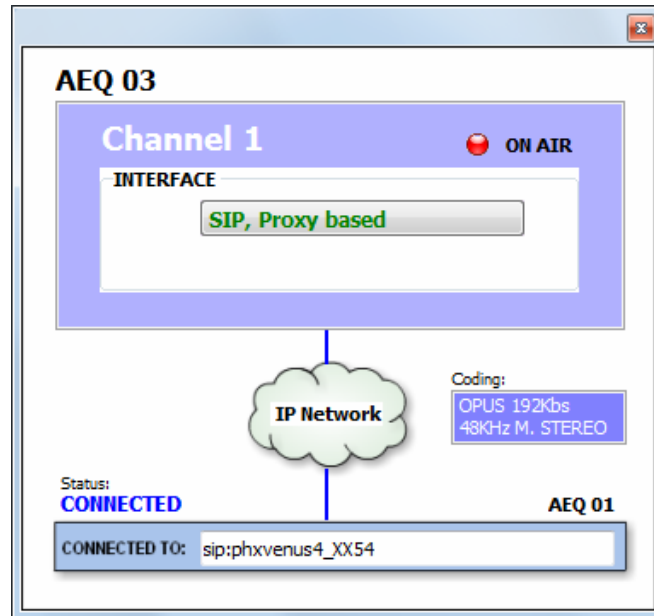
The individual codec control window is thoroughly described in chapter 6 of "AEQ ControlPHOENIX" user's manual. The screen corresponding to TALENT is as follows:



The name assigned to the device can be seen in the lower area of the window ("AEQ 03" in this example), as well as the corresponding URI.

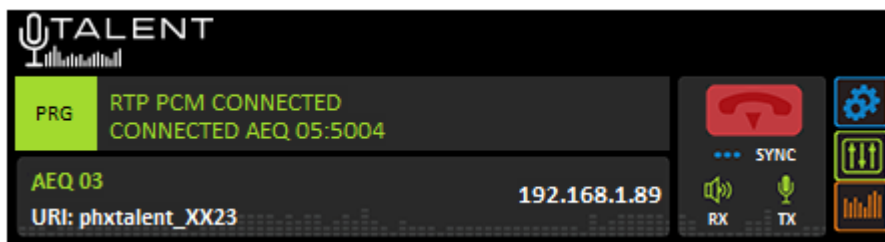
Additionally, we can see the equipment's IP address. By clicking on this IP, an Internet browser will pop up showing TALENT Web management window allowing, among other things, to update firmware and obtain real-time IP traffic statistics when the channel/s is/are connected.



The left zone shows the general status of the communications channel, (CONNECTED, OK, REGISTERING, etc), as well as the configured working mode and encoding and the remote equipment's data (IP address and port or name) in case it is connected. We can click in any of both areas in order to show a window that provides all the details of the channel we have clicked on:

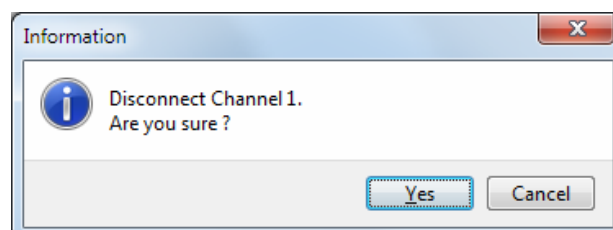


- **INTERFACE:** indicates the operating mode of that channel (RTP Raw, DIRECT SIP or Proxy SIP)
- **Coding:** indicates the coding algorithm (RTP) or profile (SIP): This section also indicates the **binary rate** (192 Kbps for example), **sampling frequency** (48 KHz, for example) and **mode** (Mono, Stereo, Dual, JStereo or MS Stereo).
- **Status:** SIP registering status for IP connections using Proxy SIP mode ("REGISTERING", "REGISTRATION_ERROR", "REGISTERED") or interface status ("OK", "CONNECTED", "CONNECTED_NO_DATA / NO_SYNC").
- **CONNECTED TO:** calling/called equipment's name or number (identifier, number or Unknown when ID is not provided) or "not connected" when there is no established communication.


On the right part, we can find the call button/indicator and the "**SYNC**" indicator, as well as two audio presence indicators for both directions: transmission ("TX") and reception ("RX").



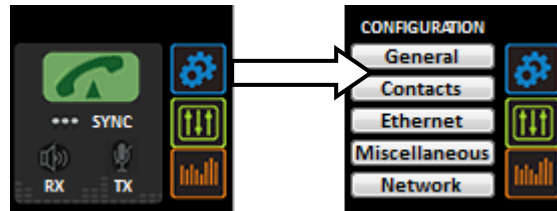
The call key changes from  to  depending on whether a communication is currently established or not. When it is green, clicking on it opens the call window (see section 6.7 of "AEQ ControlPHOENIX" user's manual). When it is red, clicking on it hangs up the current call, after asking for confirmation:



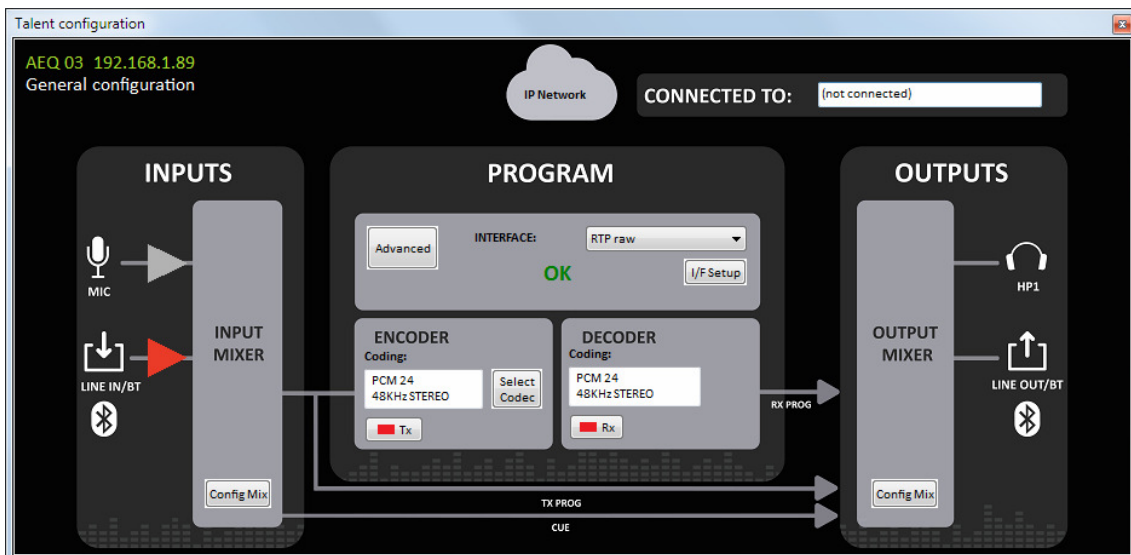
4.1.1. CONFIGURATION Menu.

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

Just click on this button again in order to close this menu.



The "**General**" option is the most important one among those associated to the configuration menu: you can configure the audio routing and levels from/to the equipment, the selected audio encoding algorithm, the interface to be used (from "INTERFACE" drop-down menu) and access to "Advanced" channel configuration and IP interface configuration ("I/F setup" button).



The general view offers a graphic representation of the audio flow inside the unit. The equipment's inputs (microphone and line / Bluetooth input), are shown at the left, entering the input mixer (that can be open by clicking on the "Config Mix" button - see paragraph 4.1.2).


This mixer outputs two buses: TX PROGRAM and CUE prelisten.

The first bus is sent to the PROGRAM block, where several aspects related to communications (communication type, ports, etc), coding algorithms, etc. can be configured.


The CUE bus is routed directly to the outputs block, as explained below.

The output block is presented at the right, where the stereo headphones and stereo line / Bluetooth output appear. PROGRAM TX and RX buses, as well as CUE, arrive to this block in order to be able to output or monitor them. The "OUTPUT MIXER" block allows for the configuration of the assignment of buses to outputs, level control and send/receive balance adjustment by clicking on the "Config Mix" button.


The amplifier symbols at the inputs are filled in red colour when they are ON air, in order to ease quick identification of the audio routing at a glance.

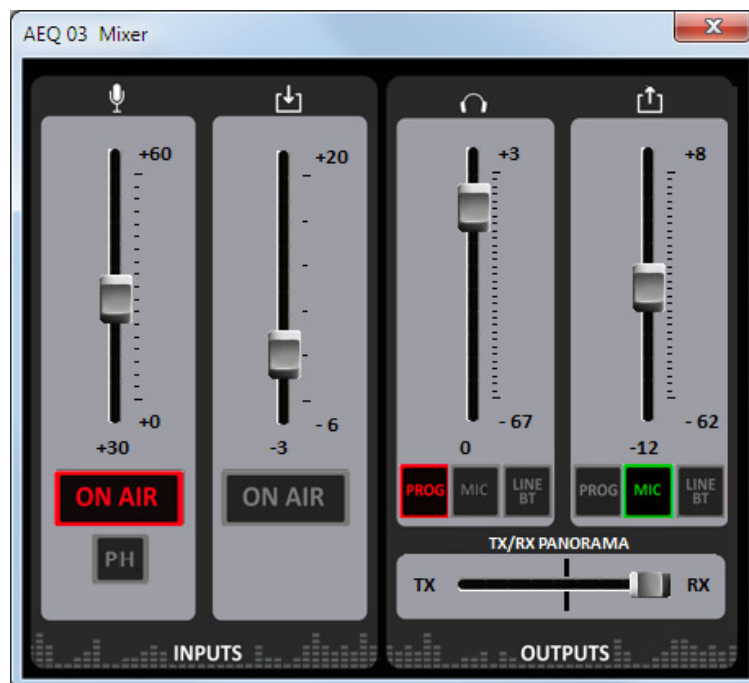
NOTE: The "Config Mix" button opens the complete mixer that controls both the inputs and the outputs, just the same as the  button located at the right side of the individual codec window, as you can see in section 4.1.2.

Other configuration menu options are "Contacts" (call book management), "Ethernet" (device IP configuration), "Miscellaneous" (several adjustments) and "Network" (other network functions parameters). Details about these menus are provided in **chapter 6** of "AEQ ControlPHOENIX" user's manual.

Just click on the  button again to close this menu.

4.1.2. Mixer control window.

Also at the right side, the  button opens a window where all parameters of the TALENT mixer can be controlled: level and gain for inputs and outputs, output routing and level, etc.



Audio can be controlled in real-time from the control PC alternatively or in parallel with the unit front panel and "AEQ TALENT PILOT" App.

MIC and Line/ Bluetooth input channels feature the following controls:


- "ON AIR" key. It will change to red color when activating the corresponding button or key, and so will the amplifier symbol at each input in the General view, simultaneously, in order to ease audio routing understanding at a glance.
- Mix fader: allows for adjustment of the corresponding input: The selected value is shown under the fader. Adjustment range is from 0 to +60dB for the microphone input and -3 to +20dB for the line / Bluetooth input.
- "PH" (Phantom) button: activates or deactivates Phantom supply for the microphone input.

The right area of the Mixer window provides control over the two outputs (headphones + Line / Bluetooth out) in a similar way: the source can be selected for each output from PROG bus, MIC input or Line / Bluetooth input.

The volume of each output can be varied by means of the associated fader. The adjustment range for the headphone output is from -67 to +3dB, while it varies from -62 to +8dB for the line / Bluetooth output.

Last, it is possible to adjust the balance between transmitted and received program signal with the "TX/RX PANORAMA" slider for the headphone and line / Bluetooth outputs currently listening to the program bus.

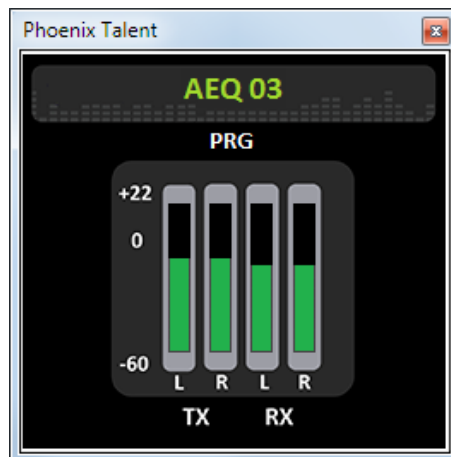
4.1.3. Vumeters window.

Last, you can access to the equipment's remote vumeters by clicking on the  button. They will appear into a floating window that you can move 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 pieces of equipment 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.

However, 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 "AEQ ControlPHOENIX" user's manual).

The vumeters represent the transmitted audio level to the L and R channels, and the received L and R channels from the communications channel.



4.2. Connection modes.

In order to establish an IP communication, first we need to choose one of the three available connection modes: "PROXY SIP", "DIRECT SIP" and "RTP Point to Point (RAW)".

We can access the IP configuration submenu by clicking on "I/F Setup". This menu is described in sections 6.1.4.2 and 6.1.4.3 of "AEQ ControlPHOENIX" user's manual.

It is important to know the details of each type of connection, so they are explained below.

4.2.1. RTP Point to Point (RAW).

This type of connection is selected when the connection over IP uses the RTP protocol only. Calls are sent by specifying the destination's IP address:port, which must be known at both sides. Obviously there is no advanced signaling protocol in this scenario and you will need to agree on audio coding algorithms at both ends and manually establish the communication in both directions.

In order to avoid that hassle (making the call from both ends, hanging up separately and agreeing on the encoding algorithm) "**SmartRTP**" can be activated in both involved audiocoders (provided that they are AEQ Phoenix units compatible with this mode). This way, the call can be established from one end only (and thus, we only need to know the IP address and ports of one of the communication ends).

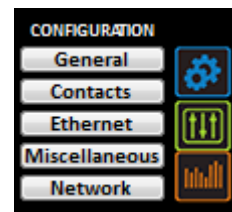
When calling in RTP mode, the destination identifier must look like this:

<IP_address>:<destination port>, for example: "**172.26.33.28:5008**"

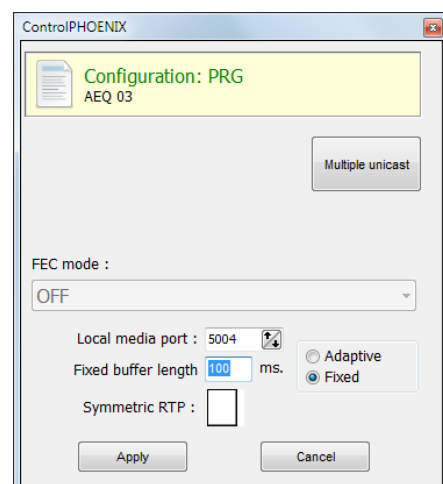
The specified destination port must match the Local port set up for the remote equipment. That is, in order to make a RTP call, we must know the IP address and local port of the remote unit even if "**SmartRTP**" is used.

However, if the destination unit is behind a router (NAT), then a port-forward rule needs to be applied on the router to re-direct the public destination specified in the destination identifier towards a certain internal IP address (the destination codec's local IP address) and port. The internal port doesn't need to be the same as the external or public port, but it usually is)

When you create a Call Book, these fields describing a contact can be modified in the Call Book that can be accessed from a codec individual control window through the "Contacts" option in "Configuration" (see section 5.1.7 of "AEQ ControlPHOENIX" user's manual).



You can access the IP configuration submenu for RTP mode (especially in order to set the "local port", where the codec channel expects the audio to be received at) by clicking on "**I/F Setup**" button, and that it is explained in section 6.1.4.3 of "AEQ ControlPHOENIX" user's manual.



The parameters to be configured are:

- **FEC mode:** this option allows you to configure whether FEC (Forward Error Correction) is used or not (there is a trade-off for a bigger binary rate). See section 4.4.
- **Local media port:** this option allows you to configure the number of the IP port where that channel decoder expects the audio to arrive at. Minimum value 1,024. Maximum value 65,534. Default value: 5004
- **Adaptive / Fixed and Adaptive buffer max/min / Fixed buffer length:** this option allows you to configure the type and size of reception buffer. See section 4.4.
- **Symmetric RTP:** when "**SmartRTP**" mode is not activated, this advanced option at least allows the user 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 audiocodec with unknown IP and/or port (because it's behind a router with NAT, for instance).

Please notice that "RTP Point to Point" is a complex configuration mode, suitable for permanent connections that some equipment may not support. Specifically, it can't be used to connect to Phoenix Mobile. However, it becomes an easy and effective way to operate with the aid of "**SmartRTP**".

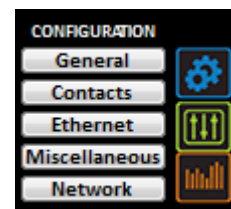
4.2.2. PROXY SIP.

This type of connection is selected when the TALENT is used working together with an external SIP server that will provide connection with remote unit through any network, even Internet, without caring about IP addresses. Both units (local and remote) must be registered in the same SIP server, which function is to maintain a database with the registered codecs, storing their connection parameters (IP address, audio ports...) in order to ease the task of making calls between them even when connected to different networks.

In order to make a call in Proxy SIP mode, you must take into account that for the **URI** or SIP identifier of the equipment in question you can use any of the following syntaxes:

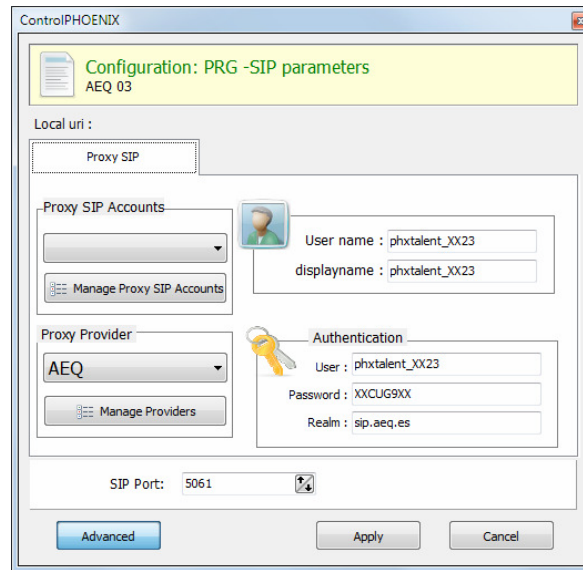
- **<unit_name>**, for instance: "phxtalent_1231" or "phoenixMaster".
- **<unit_name>@<realm_SIP_server>**, for instance: "phxtalent_1231@sip.aeq.es" o "phoenixMaster@sip.aeq.es".
- **<unit_name>@< SIP_server_IP>**, for instance: "phxtalent_1231@232.168.1.2" o "phoenixMaster@232.168.1.2" (where 232.168.1.2. is the AEQ's SIP server "sip.aeq.es")
- **<unit_name>@<SIP_server_IP>:<Port>**, when the SIP port of SIP server is not the 5060 (SIP Standard port) the identifier must include the used port, for instance: "phxtalent_1231@sip.aeq.es:5061".

When you create a Call Book, these fields describing a contact can be modified in the Call Book that can be accessed from a codec individual control window through the "Contacts" option in "Configuration" (see section 5.1.7 of "AEQ ControlPHOENIX" user's manual). In order to call a same contact using different IP modes (as defined in "INTERFACE" drop-down menu), different contact entries must be created.



You can access the IP configuration submenu for PROXY SIP mode by clicking on "I/F Setup" button, and that it is explained in section 6.1.4.2 of "AEQ ControlPHOENIX" user's manual.

- In "**SIP Parameters**" submenu you can find the following parameters (the first 3 only appear when pressing the "**Advanced**" button):



- **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. For a start we recommend you not to change the configured default User Name, "phxtalent_1231" for instance.
 - **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 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).
 - **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:
 - **User:** the "User Name" configured in Factory, "phxtalent_1231" for instance.
 - **Pwd:** the Password associated to that user.
 - **Realm:** the domain where the SIP Server is located, by default: sip.aeq.es.
 - **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.
- You can find the NAT mode selection at "**NAT Traversal**" submenu.
NAT Traversal is a set of tools used by the equipment in order to surpass the NAT (Network Address Translation) performed by some routers. We can select among several modes depending on the kind of network the unit is connected to.
TALENT offers a total of six different operating modes when traversing devices with NAT (routers, firewalls...). Each one of those modes is suitable for a different scenario. For instance, when the units that are establishing communication are in the same local network, the internal working way will be different than when it's done through the Internet.
See more details in section 4.3 of this manual.
 - The rest of options to be configured are:

- **FEC mode:** this option allows you to configure whether FEC (Forward Error Correction) is used or not (there is a trade-off for a bigger binary rate). See section 4.4.
- **Local media port:** this option allows you to configure the value of the IP port selected to transmit audio at origin over IP. Minimum value 1,024. Maximum value 65,534. Default values: 5004
- **Adaptive / Fixed and Adaptive buffer max/min / Fixed buffer length:** this option allows you to configure the type and size of reception buffer. See section 4.4.
- **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 equipment with unknown IP and/or port (because it's behind a router with NAT, for instance).

Each unit will send audio to the "Local media port" of the remote equipment automatically, thanks to the SIP signaling protocol. That signaling also accomplishes coding profile negotiation and call establishment / release from any of both sides of the communication once the remote equipment has been identified by its IP address and reached.

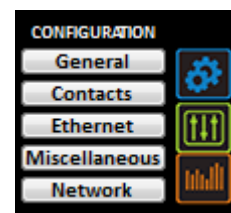
4.2.3. DIRECT SIP.

This type of connection is selected when you have a connection with SIP protocol in the signaling phase prior to connection, but without the presence of an external SIP server. It is necessary to know the IP address of the equipment you want to call in advance, but not necessarily the destination audio port.

In order to call in DIRECT SIP mode, you must take into account that for the **URI** or SIP identifier of the equipment the right syntax is "<unit_name>@<unit_IP_address>" type (for instance, "phxtalent_1231@172.26.5.57").

When the correspondent **SIP port** is not the 5060 (SIP Standard port) the identifier must include the used port. For instance: "phxtalent_1231@172.25.32.11:5061"

When you create a Call Book, these fields describing a contact can be modified in the Call Book that can be accessed from a codec individual control window through the "Contacts" option in "Configuration" (see section 5.1.7 of "AEQ ControlPHOENIX" user's manual). In order to call a same contact using different IP modes (as defined in "INTERFACE" drop-down menu), different contact entries must be created.



You can access the IP configuration submenu for DIRECT SIP mode by clicking on "I/F Setup" button, and that it is explained in section 6.1.4.2 of "AEQ ControlPHOENIX" user's manual.

- In "**SIP Parameters**" submenu you can find:
 - **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. For a start we recommend you not to change the configured default User Name, "phtalent_1231" for instance.
 - **Display Name:** editable name. This is the public name of the equipment, 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.
- You can find the NAT mode selection at "**NAT Traversal**" submenu.
 NAT Traversal is a set of tools used by the equipment in order to surpass the NAT (Network Address Translation) performed by some routers. We can select among several modes depending on the kind of network the unit is connected to.

Phoenix TALENT offers a total of six different operating modes when traversing devices with NAT (routers, firewalls...). Each one of those modes is suitable for a different scenario. For instance, when the units that are establishing communication are in the same local network, the internal working way will be different than when it's done through the Internet.

See more details in section 4.3 of this manual.

- The rest of options to be configured are:
 - **FEC mode:** this option allows you to configure whether FEC (Forward Error Correction) is used or not (there is a trade-off for a bigger binary rate). See section 4.4.
 - **Local media port:** this option allows you to configure the value of the IP port selected to transmit audio at origin over IP. Minimum value 1,024. Maximum value 65,534. Default value: 5004
 - **Adaptive / Fixed and Adaptive buffer max/min / Fixed buffer length:** this option allows you to configure the type and size of reception buffer. See section 4.4.
 - **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 audiocodec with unknown IP and/or port (because it's behind a router with NAT, for instance).

Each unit will send audio to the "Local media port" of the remote equipment automatically, thanks to the SIP signaling protocol. That signalling also accomplishes coding profile negotiation and call establishment / release from any of both sides of the communication once the remote equipment has been identified by its IP address and reached.

4.2.4. Sending audio to multiple destinations: Broadcast, Multicast and Multi-unicast.

It is possible to send the same audio RTP stream to several different destinations in "RTP raw" mode (see section 4.2.1). There are several possibilities to do so (see "AEQ ControlPHOENIX" manual):

a) Broadcast: the audio stream can be sent to all the devices 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 pieces of equipment should have "local media port" set up to this same port so they are able to receive the RTP stream.

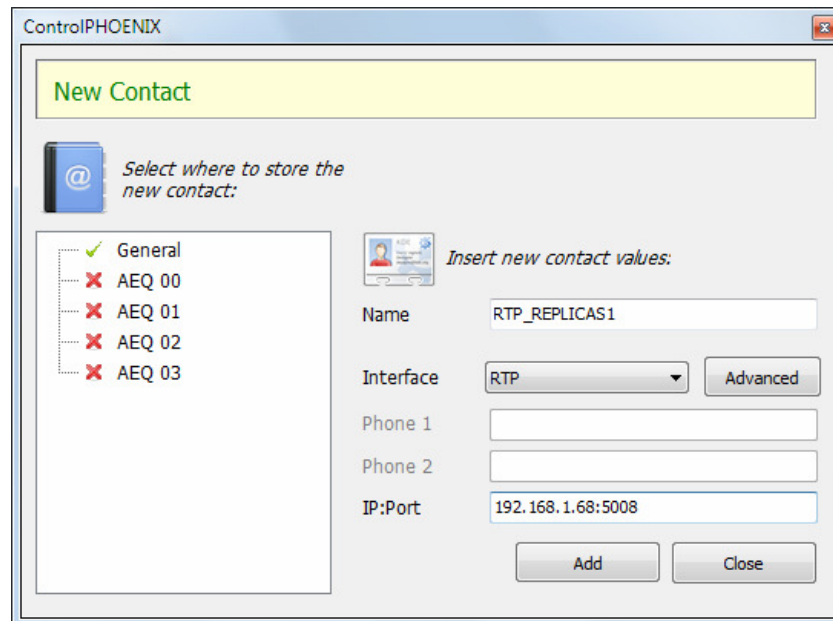
This mode is not recommended for big networks and is usually blocked by the switches and routers, so its use is restricted to small, well managed local area networks.

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 devices 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. TALENT implements IGMP (Internet Group Management Protocol) in order to subscribe to multicast group. 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: TALENT 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

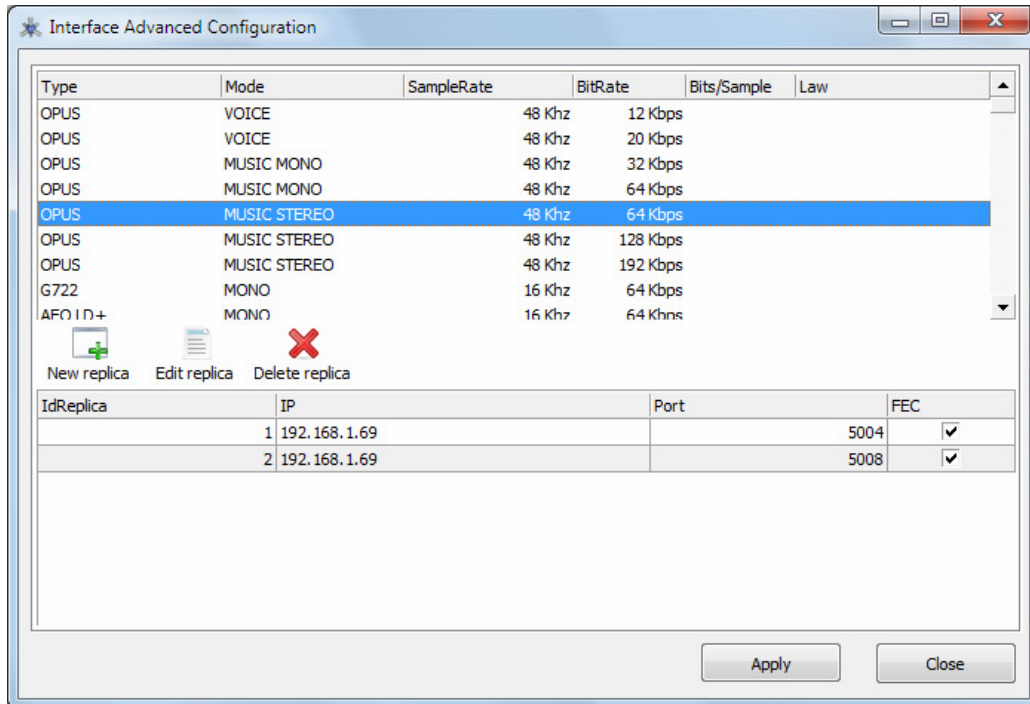
That parallel streams or "replicas" at nothing more than IP address + port pairs, where audio copies are to be sent normally. When a contact is created or edited, it is also possible to specify whether particular replicas use FEC (forward error correction) if it is enabled for that channel, or disable it for certain streams (because they use stronger links, for example). If the above list is empty, the audio stream will be sent to the IP + port specified when making the call (main address in the contact).

In order to send replicas, a new advanced contact must be created first in the General agenda where a main IP address and port is specified and a list of additional replicas is provided. In order to do that, click on the "Contacts" button at the top menu bar, select "New Contact" (make sure that the agenda selected at the left column is "General" and proceed with the creation of the new contact:



Give the new contact a name (i.e: "RTP_REPLICAS1"), select RTP as the interface mode, specify the main destination IP address and port (192.168.1.68:5008, respectively, in this example) and click on the "Advanced" button to specify the selected coding method and add replicas to the list.

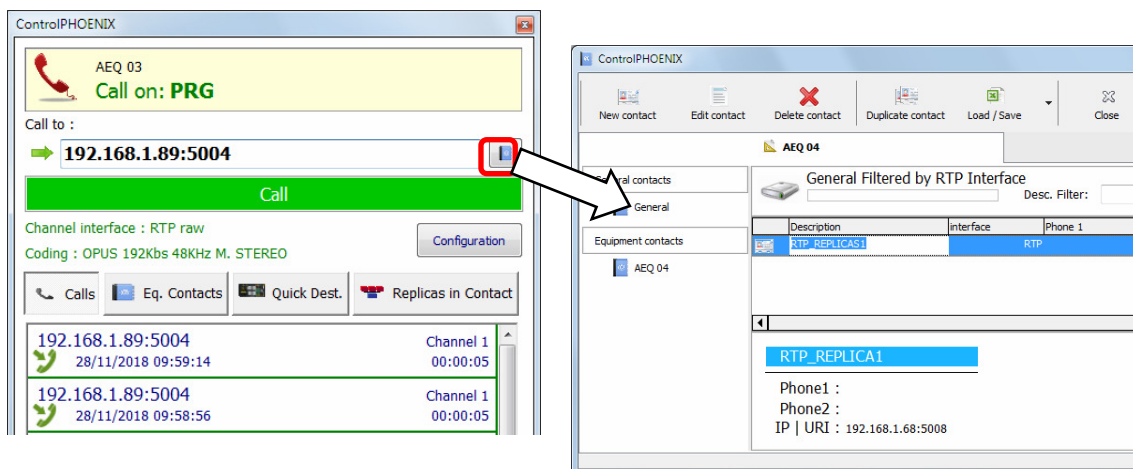
The OPUS MUSIC STEREO 48 KHz / 64 Kbps mode will be used in this example, configuring the contact to issue replicas to 2 different IP/Port pairs (we have used the "New Replica" button in order to do that).



Detail of the creation of RTP replicas within a General agenda contact

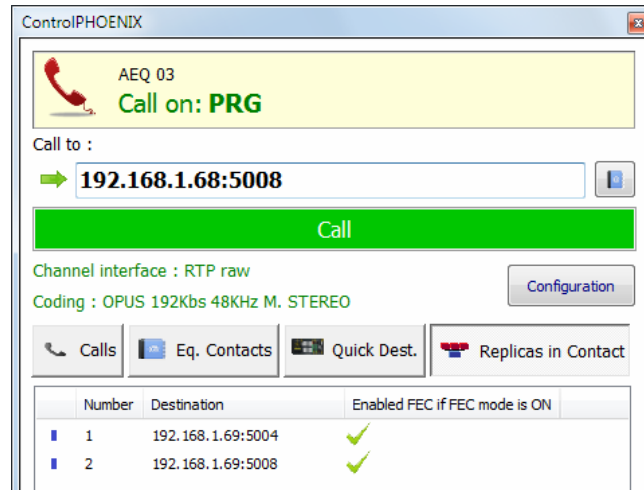
Please check chapter 5.1.7.1 of "AEQ ControPhoenix" User's manual for more details.

We need to select this contact that includes replicas from the call window when making a call.



Once the available contacts are displayed, select the one we are interested in ("RTP_REPLICAS1" in this example), by double-clicking its name, and we will be returned to the call window.

At this moment we are able to check that the specified main IP address and port fields fill the "Call to" field, and we can also check that the replicas are going to be loaded into the unit, by clicking on the "Replicas in Contact" before actually making the call:

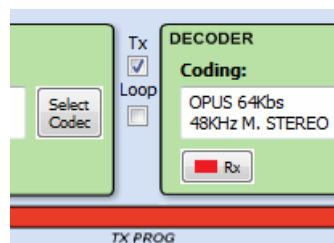


Once we click on the green "Call" button, the unit will start emitting the audio streams to the main address and to the specified replica addresses.

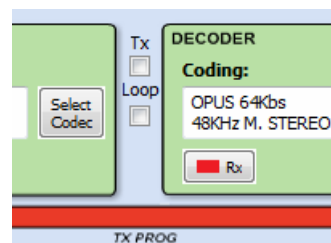
TALENT will stop sending replicas as soon as the call is manually hung. The list will be erased from the unit and the only way to send them again is to call selecting the same contact.

However, if the Permanent Call mode is activated and there is a mains cut, for example, the unit WILL send all configured replicas when rebooting.

NOTE 1: When audio is transmitted to several destinations, it can be received from only one of them (or none) In order to establish which of the units sends the audio back, and only in RTP Raw mode, a control will appear in the general configuration window that allows for the activation / deactivation of the transmitted stream to the IP channel. Make sure that only one of the units connected to the multicast transmitter has this checkbox activated.



Channel transmission active



Channel transmission disabled

NOTE 2: Advanced contacts (those allowing specification of the communication mode, coding algorithm/profile, replicas, etc.) can only be stored in the General call book, which 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 (even if it is not the main receiver).

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

4.3. NAT TRAVERSAL.

NAT Traversal is a set of tools used by the equipment in order to overcome the issues caused by NAT (Network Address Translation), typically performed by routers. We can select among several modes depending on the kind of network the units are connected to.

For IP connections using SIP protocol, TALENT offers a total of six different operating modes when traversing devices with NAT (routers, firewalls...). Each one of those modes is suitable for a different scenario. For instance, when the units that are establishing communication are in the same local network, the internal working way will be different than when it's done through the Internet.

Four of the six modes are automatic (AUTO 1 - AUTO 4), another one is manual (MANUAL - router configuration) and the last one (OFF - there is no NAT) is used when no devices with NAT are crossed (the unit is in a local network or connected to the Internet with a single workstation router). In automatic modes the unit tries to find out its public IP and ports without the user help, while in manual mode the unit gets those data directly from user (and user gets it from network administrator).

Due to the technical complexity inherent in most of the parameters involved in this NAT TRAVERSAL menu and the importance that any modification has in the final operation of the unit, we recommend that only highly qualified personnel in possession of all the technical documentation and manuals work on this NAT configuration menu. For additional information, see **APPENDIX B5**.

The NAT traversal options of a codec are accessed by following this sequence from the involved individual codec control window: "Configuration" → "General" → "I/F Setup" → "NAT Traversal".

Next we will describe the operation without NAT and the other five modes supported by TALENT.

4.3.1. Operation without NAT: "OFF (there is no NAT)".

The unit uses no mechanism to traverse devices with NAT. 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).

4.3.2. Manual NAT: "MANUAL (router configuration)".

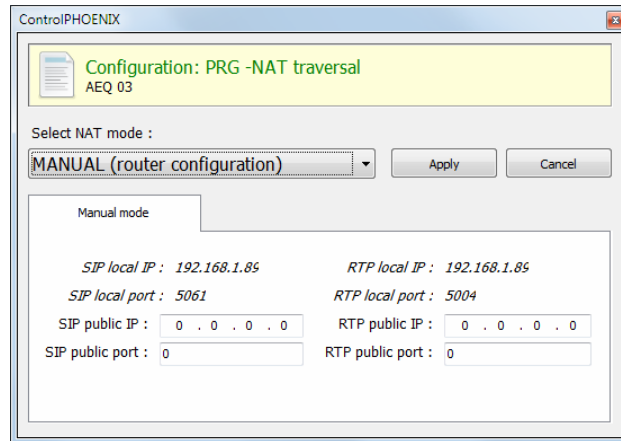
This mode will be used when the unit is connected to a local network with shared Internet access, through a router that will work as NAT (Network Address Translation). In order to use this mode no DHCP must be used and you need to have access to router configuration (and the knowledge to do it) or to the Network Administrator that will give us some data to be configured in the unit and configure the router to open and redirect some IPs and ports (port forwarding).

The procedure is described in detail in the following Application Note (we recommend you to read it when you decide to use this working mode):

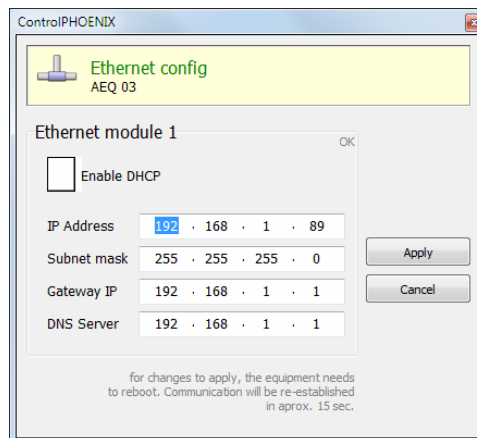
AEQ PHOENIX AUDIOCODECS. APPLICATION NOTE 0 - C

(Connecting AEQ Phoenix units through Internet. Complex scenario configuration. Through LOCAL network(s), DHCP not used, manual NAT. Making use of AEQ Proxy SIP).

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



1. **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. For instance [192.168.1.89](#). It can be set in order to adapt it to network necessities in menu: "Configuration" → "Ethernet".



2. **SIP LOCAL PORT:** read-only parameter that tells you the port of the IP interface of the unit used for SIP signalling, 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.
3. **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. For instance [212.170.180.177](#)
4. **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. For instance: 8001.
5. **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: for instance [192.168.1.89](#)
6. **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. Usually the shown port is 5004.

7. **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. For instance [212.170.180.177](#)
8. **RTP PUBLIC PORT:** parameter that will tell the Phoenix ALIO 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. For instance: 8002

In the aforementioned note the detailed configuration and need of port-forwarding is explained.

4.3.3. AUTO 1 (local network audio).

This mode will be used primarily when two units that are in the same local network need to communicate with each other, when the Proxy SIP is on the Internet and it's the one provided by AEQ (sip.aeq.es).

4.3.4. AUTO 2 (local network audio).

This mode will be used primarily when two units that are in the same local network need to communicate with each other, when a Proxy SIP on the Internet is used, it's not the one provided by AEQ (sip.aeq.es) and only if the AUTO1 mode doesn't work properly.

4.3.5. AUTO 3 (audio over internet).

This mode will be used mainly when you wish to put two units in communication with each other through the Internet, working with no Proxy (DIRECT SIP mode) or using the Proxy SIP provided by AEQ (sip.aeq.es) (PROXY SIP mode). The two configuration parameters available on screen for this mode are:

1. **STUN SERVER:** parameter that tells the unit the STUN server that will be used. On the Internet there is multitude of public STUN servers. By default, the IP address of stun.sipgate.net server is configured: 217.10.68.152.
2. **STUN PORT:** parameter that tells the unit the STUN server port assigned by the administrator. By default: 3478.

NOTE: in this mode the TALENT behaves in the exact same way as the Phoenix Mobile unit when it is using a STUN server.

4.3.6. AUTO 4 (audio over internet).

This mode is equivalent to AUTO3 but it will be used the SIP server is not the one provided by AEQ and there are problems with AUTO3 mode. The configuration parameters are the same as for AUTO3 (STUN server specification).

4.4. FEC modes and reception buffer configuration.

- **FEC error correction mode.** 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 and 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.

NOTE: please activate FEC only when advised by AEQ Technical Support, as improper usage may cause more trouble than simply not using it, depending on the network characteristics.

- **Adaptive / Fixed:** you can set up 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. We always recommend starting with a fixed configuration
- **Adaptive Buffer max/min / Fixed buffer length:** these are the maximum y minimum size of the reception buffer. When it is defined as adaptive, TALENT will start to adjust it from these values as the network's transmission quality allows. If it is defined as FIXED, this 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, especially if the buffer is set to FIXED mode.

SOME RECOMMENDATIONS:

In order to help you select the best option for each application, we recommend to use a Fixed buffer, with a low value (around 100ms) in applications where optimal audio quality is the main concern (mainly when using PCM modes in suitably sized networks). If the received audio quality is as expected, and the network allows for it, you can continue adjusting the buffer to lower values in order to minimize delay, until you find that audio is compromised (as the buffer size reaches the network maximum jitter value). At this point, just increase the buffer a little bit to have some margin.

In high-quality PCM connections, and if the network allows for it, you can start using highest quality modes (48 KHz, 24 bits, mono or stereo only if required), and if you can't obtain the desired quality and/or stability (no noises present) and good delay, you can lower quality progressively until, for example, 16 bit (CD quality audio).

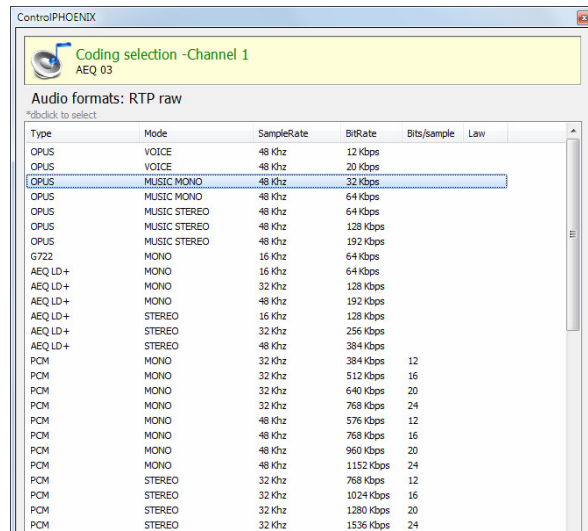
On the other hand, for applications where lowest possible delay is the main goal, but transparent audio is not necessary (for example, in voice connections with commentators), it is better to select the Adaptive Buffer mode, starting from a 1000 ms maximum size, approx. If the network is not too bad, the unit won't increase the buffer to highest values from the network's jitter value, and it will try to minimize delay continuously. Please not that if the network has very variable delay, the adjustments required to increase or decrease the buffer size can produce noticeable effects in the received audio, so this method is not usually recommended for a start.

4.5. Coding algorithm selection.

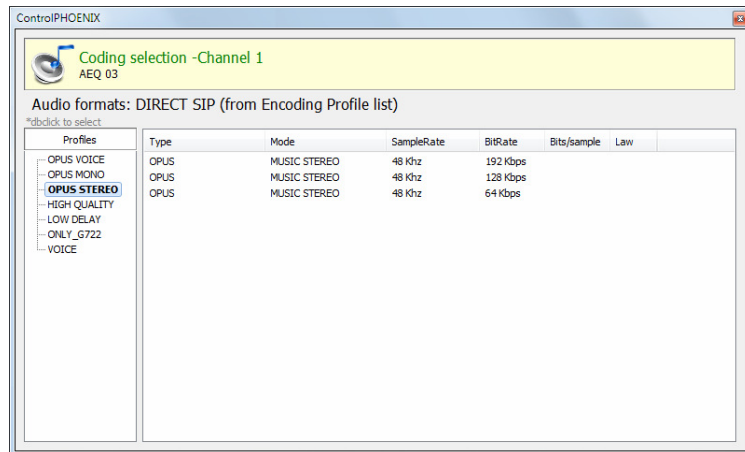
See section 6.1.3.1 ("Coding selection") at "AEQ ControlPHOENIX" user's manual.

Although AEQ recommends the use of OPUS coding algorithms for most uses (several different modes are available to match almost any need), a coding selection window can be accessed by clicking on the "Select codec" button located inside the "ENCODER" area of the general configuration window, where both OPUS modes and others provided for compatibility can be found.

Note that the "DECODER" will be automatically configured for the same coding algorithm and mode.



However, when the interface is in any of the SIP modes ("DIRECT SIP" or "PROXY SIP"), particular coding algorithms can't be selected and connection "profiles" will be defined instead, containing one or more modes. The reason for this is that SIP allows for the participants in a communication to negotiate the coding algorithm, so the one to be finally used will be limited to those includes in the selected profile.

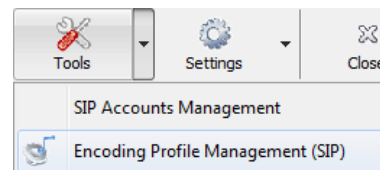


This possibility allows for the configuration of the parameters associated to the coding to be used in Audio over IP networks basing on SIP protocol (Proxy SIP and Direct SIP modes). This option simplifies the selection of the algorithm to be used in a communication, because most of the codecs have several tens of encoding algorithms in order to have the higher compatibility with third-party devices.

When a communication is established using SIP signalling, the codec negotiates the use of the first compatible encoding algorithm included in a list called SIP CODEC PROFILE. That's why you should put these algorithms in order of preference.

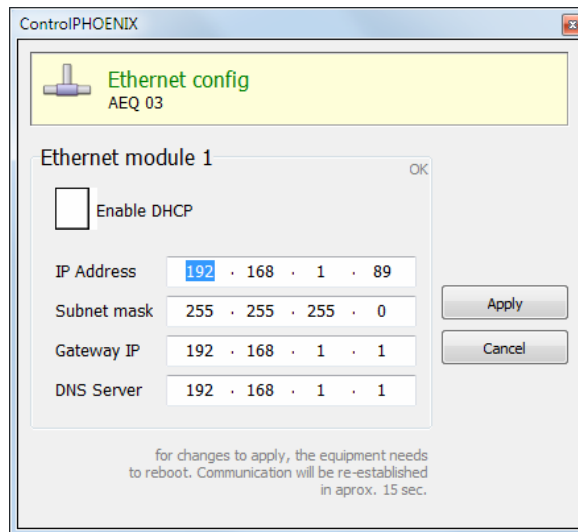
Each one of the stored entries includes an alphanumeric identifier and a list of algorithms to use, organized in order of preference. There are several preset profiles in the unit, grouped by delay, quality... etc.

Profiles can be added, modified or deleted in the "Encoding Profile Management (SIP)" screen, accessible from the "Tools" menu in the upper Menu bar, described in paragraph 5.1.9 of "AEQ ControlPHOENIX" user's manual.



4.6. Ethernet Port configuration.

The "**Ethernet config**" menu allows for the configuration of the unit's Ethernet interface IP parameters.



The parameters to be configured are:

- **Enable DHCP:** enables the activation or deactivation of the automatic IP address, mask and gateway configuration. There must be a DHCP server in the network the unit is connected to in order for this option to work. When "Enable DHCP" is active, the following parameters will be filled automatically; when "Enable DHCP" is not active, 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.
- **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.

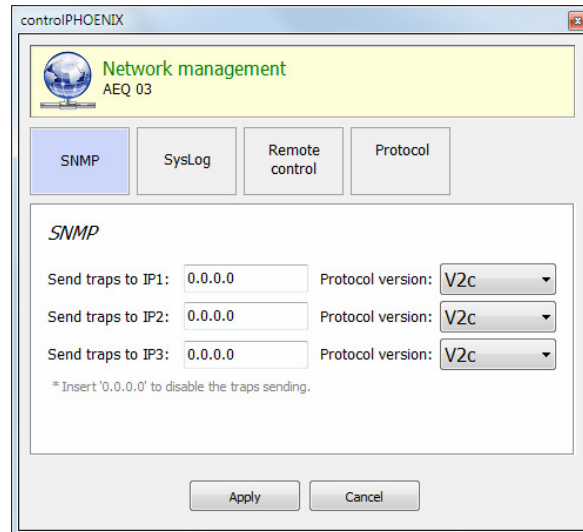
Once those parameters are configured, and after clicking on the "Apply" button, a confirmation window will appear. After confirming, the equipment reboots and the communication re-establishes in approximately 15 seconds.

If you have any doubts, please consult your IT network technician or directly contact AEQ or authorized distributors' technical support department.

4.7. SNMP configuration.

This unit can be remotely managed using SNMP (Simple Network Management Protocol) using one of the many software clients available in the market, even for free. SNMP allows for monitoring of the status of several pieces of equipment from very diverse manufacturers and natures, as well as elaborating reports, generate e-mail alerts, etc.

You can access the configuration menu in "Configuration" → "Network".



AEQ Phoenix audiocodex family can connect to **up to 3 SNMP clients**, installed in remote PCs, by simply configuring their IP addresses in the "SNMP" tab of previous menu. Once one or more SNMP clients are connected and the corresponding "MIB" descriptive file has been loaded (it can be downloaded from the equipment's Web Interface: see section 7.6 of this manual), the audiocodex will send/accept different types of information to/from each client:

1. Alarms (TRAPS): Provide information relating to certain equipment's events that can be considered as alarms. They usually have two possible states: Active or Inactive. The list of alarms defined for TALENT is as follows:

- phxCh1NoTxAudioAlarm (Audio detection event at Channel 1's input)
- phxCh1NoRxAudioAlarm (Audio detection event at Channel 1's output)
- phxCh1NoAudioSyncAlarm (Sync event at Channel 1 while connected)
- phxCh1CallEndAlarm (Call ended on Channel 1 due to incoming RTP traffic loss)
- phxOtherAlarm (Other alarms, see name) -> Fail to register in SIP PROXY server, activation / deactivation of BACKUP interface for Ch1
- coldStart alarm (Starting from unit off. This is a standard SNMP alarm, it appears only one time and has no activation or deactivation).

These alarms are sent whenever they change, but we can choose from the SNMP client which ones are shown / treated and which not.

The first 4 audio alarms are configurable and they are activated whenever the incoming or outgoing (depending on the particular alarm) audio level is below a certain threshold for a given time (configured by means of the remote control software or by issuing the following configuration commands). For example, alarm "phxCh1NoTxAudioAlarm" will become Active whenever audio from ch1 input has a level below the threshold defined by "phxCh1TxAudioThreshold" during a longer time than specified by "phxCh1TxAudioInterval".

2. Configurations (SET): adjustments related to some of the above defined alarms (the SNMP client will configure them by means of "SET" commands, although in the case of Phoenix units, they can also be modified by means of "AEQ ControlPHOENIX" remote control software):

- phxCh1TxAudioThreshold (Audio threshold for channel 1's input)
- phxCh1TxAudioInterval (Audio interval for channel 1's input)
- phxCh1RxAudioThreshold (Audio threshold for channel 1's output)
- phxCh1RxAudioInterval (Audio interval for channel 1's output)

3. Information messages (GET): showing a status, they don't arrive spontaneously or are activated/deactivated like the Alarms, but they are requested by the SNMP client by means of "GET" messages:

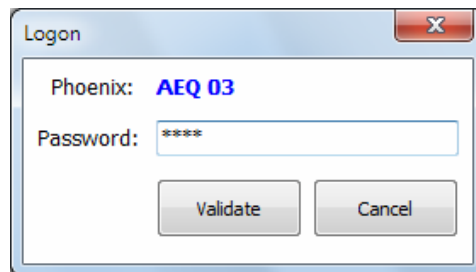
- phxCh1Status (Channel 1's status) --> (OK, CONNECTED, CONNECTING, DISCONNECTING, CALLING, NO LINK, REGISTER ERROR, REGISTERING, CALL ERROR)
- phxCh1TxAudio (Ch1's audio input indicator status). Indicates the status of the virtual LED indicating audio presence, according to the corresponding configured parameters.
- phxCh1RxAudio (Ch1's audio output indicator status). Indicates the status of the virtual LED indicating audio presence, according to the corresponding configured parameters.
- phxCh1OnAir (Ch1's ON AIR option activated or not)
- phxCh1Synced (Ch1's audio synchronized or not)
- phxCh1BackupInterfaceActive (Whether or not Ch1's backup interface is being used on a call)
- mib2.system (sysUpTime, sysContact, sysDescr, sysServices...): Standard SNMP commands indicating things such as equipment's turn on time, etc. For more information, please check MIB-II specification in RFC1213 (<http://tools.ietf.org/html/rfc1213>)

For more information, please consult section 7.6 of this manual and section 6.5.1 of "AEQ ControlPHOENIX" application manual.

4.8. Password configuration.

Version **2.2.1.1** (or higher) of the application allows you to configure a numerical **password** of up to 4 digits required in order to be able to control the unit, by clicking on "User Password" checkbox (in "Syslog" tab of "Network management" menu).

That password will be requested whenever you try to **add** the unit to the application, whether it's done by autodiscovery or manually (once the unit is added, the password won't be requested again even if the software is closed and reopen, 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).

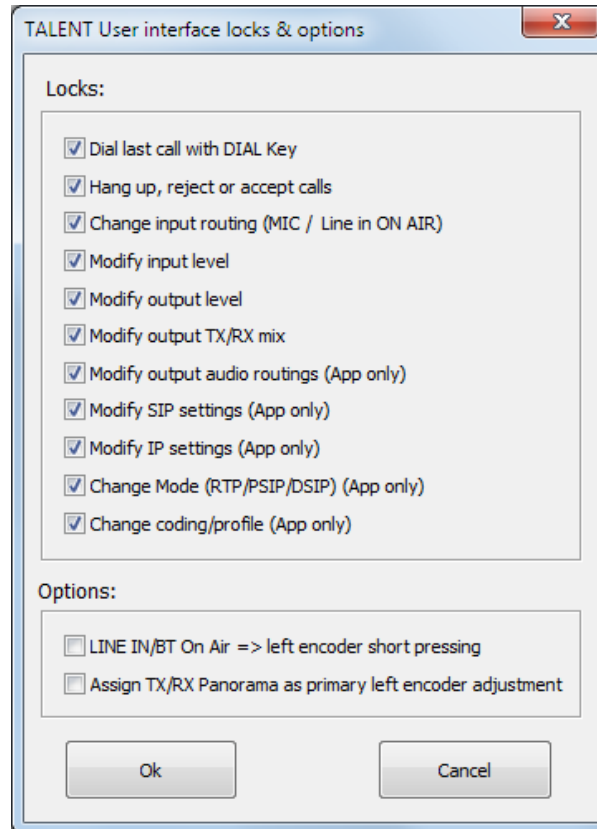


For more information, please consult section 6.5.3 of "AEQ ControlPHOENIX" application manual.

4.9. User interface lock.

It is possible to lock certain functions of the physical user panel from "**AEQ ControlPHOENIX**" software. The only way to recover local control of the disabled functions is by connecting the unit using the control application and re-enabling them. It is also possible to modify 2 options of the local control interface.

By means of "**User interface locks & options**" option in the "**Miscellaneous**" menu (see section 6.4.9 of "AEQ ControlPHOENIX" user's manual), the window that allows you to enable/disable those permissions will be accessed:



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".

These settings are stored on the device, so if it is controlled from another PC, they will still apply.

These are the groups of actions that can be enabled/disabled:

- **"Dial last call with DIAL Key"**: allows or forbids the user to call the latest configured destination using the unit's user panel keys.
- **"Hang up, reject or accept calls"**: allows or forbids the user to hang up, reject or accept calls from the unit's interface.
- **"Change input routing (MIC / Line in ON AIR)"**: allows or forbids the user to change Input Routing from TALENT's user panel. The restriction affects all 2 inputs.
- **"Modify input level"**: allows or forbids the user to modify the inputs send level by turning the associated rotary encoder.
- **"Modify output level"**: allows or forbids the user to change the headphone and line output level by turning the associated rotary encoder.
- **"Modify output TX/RX mix"**: allows or forbids the user to adjust the ratio of transmission and reception of the corresponding output by turning the associated TX/RX potentiometer.
- **"Modify output audio routings (App only)"**: allows or forbids the user to change from "AEQ TALENT PILOT" App the outputs audio routing.
- **"Modify SIP settings (App only)"**: allows or forbids the user to change from "AEQ TALENT PILOT" App the SIP configuration of the unit.
- **"Modify IP settings (App only)"**: allows or forbids the user to change from "AEQ TALENT PILOT" App the IP configuration of the unit.

- **"Change Mode (RTP/PSIP/DSIP) (App only)":** allows or forbids the user to change from "AEQ TALENT PILOT" App the communication mode (RTP, Proxy SIP or Direct SIP).
- **"Change coding/profile (App only)":** allows or forbids the user to change from "AEQ TALENT PILOT" App the encoding or profile to be used in communication.

On the other hand, the options of the local control interface that can be enabled/disabled are:

- **"LINE IN/BT On Air => left encoder short pressing":** If this option is activated, the ON AIR of the input line will be activated with a short press of the left encoder. By making a long press, we will go to the alternative level adjustment mode (usually TX/RX).
- **"Assign TX/RX Panorama as primary left encoder adjustment":** If this option is activated, the role of the primary and alternate settings of the left encoder is swapped. That is, if we simply turn it, we will adjust TX/RX Panorama, and if we press, we will adjust the MIC gain.

5. CONFIGURATION AND OPERATION OPTIONS FROM "AEQ TALENT PILOT" REMOTE CONTROL APPLICATION.

"AEQ TALENT PILOT" App, available for smart devices running Android and iOS operating systems, allows for advanced local control of AEQ TALENT audiocodec (the unit must have CPU firmware version **9.30** or above). However, for exhaustive configuration "AEQ ControlPHOENIX" software must be used, available for Windows PC platforms.

5.1. Downloading "AEQ TALENT PILOT".

For Android devices:

- 1) Search for "AEQ TALENT PILOT" at Play Store.
- 2) Select it to download and install the App.



For iOS devices:

- 1) Search for "AEQ TALENT PILOT" at Apple's App Store.
- 2) Select it to download and install the App.

5.2. Running "AEQ TALENT PILOT" for the first time.

Before opening the App, turn your device's Bluetooth ON. After some seconds, the following devices should be detected:

- AEQ_TALENT_<serial number>
- AEQ_TALENT_BLE_<serial number>

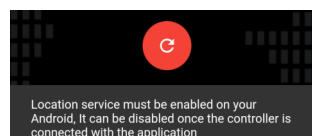
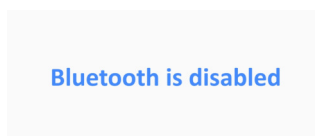
The first one is used to exchange audio signals. This one can be paired to the same device running "AEQ TALENT PILOT" or to a different one. Once audio has been linked, AEQ Talent's blue Bluetooth light should stop flashing, and become steadily illuminated.

The second device is used for remote control, and you don't need to pair it. The App will automatically detect it.

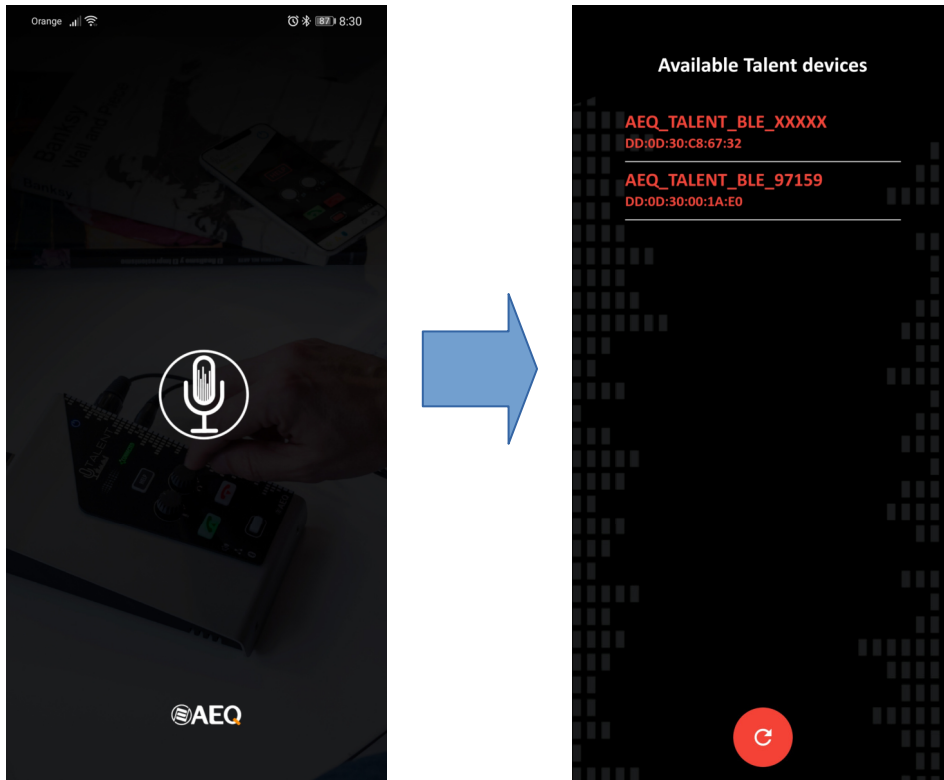
Once you have made sure that both devices are visible (you don't really need to check this every time you use the App), next run "AEQ TALENT PILOT" App.

IMPORTANT NOTE: In order that the App works correctly and the list of TALENT devices appears, the location of the phone must be activated. Additionally, the first time the App is executed, it will request permission to access the device's location while it is running. This doesn't mean that PILOT App or AEQ will use this information in any way, but it is an internal Operating System's requisite to be able to interact with the Bluetooth BLE (low energy) interface incorporated into TALENT.

The application will show an initial splash screen. If Bluetooth is not active in the smart device, a warning will show up informing about this issue; likewise, if the location of the phone is not activated, a warning about it will appear.



Otherwise, a list including all the AEQ TALENT devices found will be displayed after a few seconds. You can always touch the red Refresh button at the bottom of the "Available Talent devices" screen if your TALENT is still not found or you forgot to turn it on. In case that several TALENT codecs are at reach, simply touch on the one you want to control (make sure that the serial number matches the one written at the back label of your device).



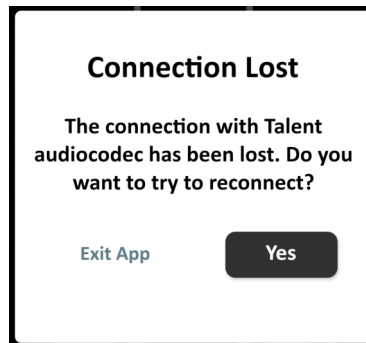
Once the App has linked to TALENT, the red light in the Bluetooth symbol of the device will be illuminated. So if the audio link is also paired, this indicator will have a pinkish color

5.3. Main Screen.

Once the desired device is found and selected, after a few seconds the main App screen will be presented:




If at a certain moment the connection with the device is lost, an informative pop-up appears that allows you to try to reconnect with the device (by pressing "Yes") or to "Exit App":






The main screen tries to resemble the actual device's front panel appearance, so it is easier to understand and use. It is divided into three differentiated areas.

5.3.1. Upper area.

This area, that is fixed and always displayed, includes the App logo as well as some basic information about the system:

- LINK indicator:** 

This symbol indicates that the TALENT device is currently linked to the App. If this symbol turns off, communications with it has been lost (for example, if the smart device is too far away, more than approx. 5 meters). The App will try to re-connect and, if impossible, it will give the user the possibility to retry or leave.
- CONNECTED indicator:**  

This indicator shows the status of the currently established communication. It can be off, on or blinking. When it is off, the communications channel is idling (no communication is attempted). When it is ON, there is a bi-directional communication successfully established. However, when it is blinking, it indicates that a call has been sent but there is no returning IP traffic, and this may simply mean that the communication is unidirectional or, more commonly, that the call hasn't been properly established with the remote end.
- VU-METER:** 

It shows the program send level in only 4 states (off, green, green+yellow, or green+yellow+red), just as it appears on the device's front panel. Level will be displayed in this vu-meter only if at least one of the inputs (MIC or LINE IN/Bluetooth) is ON AIR.

5.3.2. Main area.

This area shows a representation of the controls that is similar to the device physical controls. In fact, these controls operate in parallel with the actual ones, and touching any of them execute the corresponding actions in a coherent way.

This part of the screen changes as we access the different menus.

These are the controls available in the main area:

- **HELP button:**



Once a remote control connection has been successfully established with one or more instances of "AEQ ControlPHOENIX" software running on local or remote PCs, if this button is touched, the effect will be the same as touching the corresponding TALENT's physical HELP key: a pop-up window will appear in the software warning the operator that we are in need of assistance. The request can be cancelled at any time by means of a short touch.

NOTE: As extra to the physical key, where the "HELP" key can also activate / deactivate the Bluetooth module when pressed for a longer time, the only function of this button in the App is to ask for assistance.

- **Audio inputs control:**



This rotary encoder provides access to the audio inputs configuration screen (see section 5.5.1 of this manual). The indicator located down at its right indicates, when illuminated red, that the auxiliary / Bluetooth input is ON AIR, thus being sent to program.

- **Audio outputs control:**



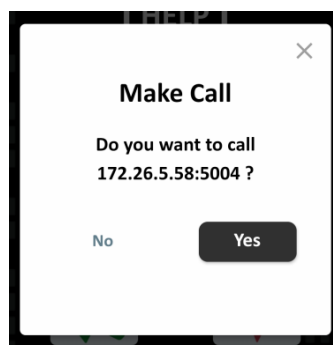
This rotary encoder allows for access to the audio outputs configuration screen (see section 5.5.2 of this manual). The indicator located down at its right indicates, when illuminated red, that the microphone is being sent to the auxiliary / Bluetooth output.

- **CALL button:**



By briefly touching this button, the Call menu is accessed (see section 5.4 of this manual). However, when pressing it for a longer time, we will launch a call to the destination that was last selected (either from the different call menu options or externally using "AEQ ControlPHOENIX").

Before the call is actually made, a pop-up window will appear, informing about the programmed destination and asking for confirmation:



- **HANG UP button:**



By briefly touching this button, the currently established call (if any) will be hang up after providing confirmation.


- **ON AIR button:**





This key sends the microphone to Program (the MIC is sent when the button is red).

5.3.3. Lower area.

The lower screen area is fixed, just as the top one. It includes several device status indicators, as well as two buttons to navigate through the application.

- Received audio indicator:** 

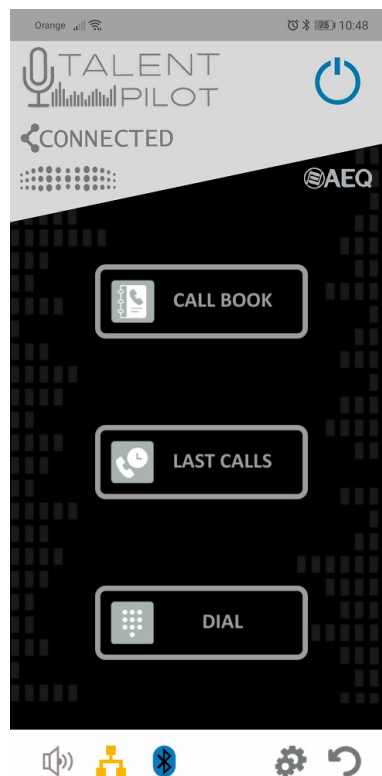
It indicates presence of received audio, according to the default levels (threshold: -57dBFS, timeout=2 secs), which can be of course modified from "AEQ ControlPHOENIX" PC software. It is required that a communication is active for this indicator to become active.
- IP connection ready indicator:** 

This indicator tells us whether the channel is ready to make a call. In RTP mode, this means that there is an Ethernet link active on the LAN port and that IP configuration is valid (it was either manually assigned or obtained from a DHCP server). In Proxy SIP (PSIP) mode, it also implies that registration on the configured SIP server is successful. If this indicator blinks, the connection is not fully ready (when in RTP mode, we have link but no valid IP configuration, or when in PSIP mode, registration is not complete).
- Bluetooth indicator:** 

When the Bluetooth symbol is blue, it means that TALENT is linked for audio (with this or any other smart device). Note that if the symbol is blue in the App, it will be pink on the device, since there will be control and audio connections simultaneously.

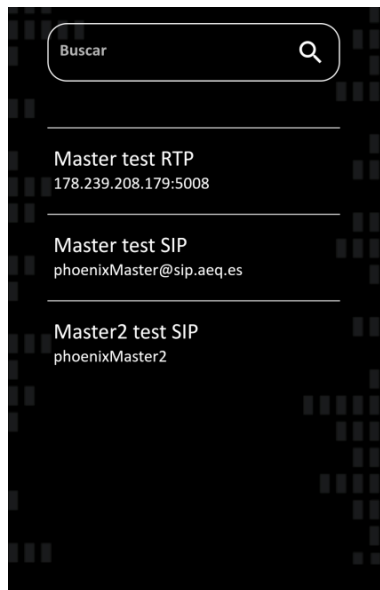
5.4. Call menu.

When the CALL button is briefly touched, the main area will display the CALL menu, offering us three ways to define the destination of a communication:



Three buttons can be found in this screen:

- **CALL BOOK:** Opens a call book stored in the TALENT device.



The management of this call book, which is saved into TALENT's internal non-volatile memory, is carried out by means of "AEQ ControlPHOENIX" PC software.

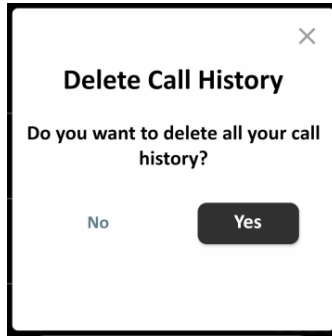
Contacts include a name and its dialling information. It is under the user's responsibility to choose adequate contacts according to the communications mode selected at each moment (Proxy SIP, RTP, Direct SIP), as the call syntax varies from one to another.

When one of the contacts is touched, a pop-up window will appear asking us whether we want to select it as the next destination or not.

- **LAST CALLS:** The App shows a list with the latest calls initiated from it and the calls received when the App was open.

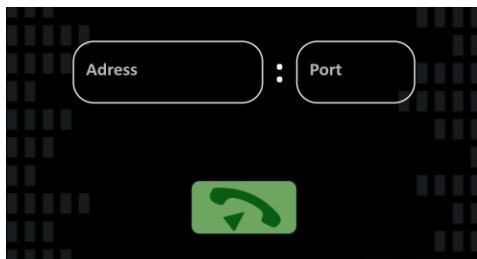


We can erase the App's call history by using the "Clear History" button, which will ask us for confirmation:

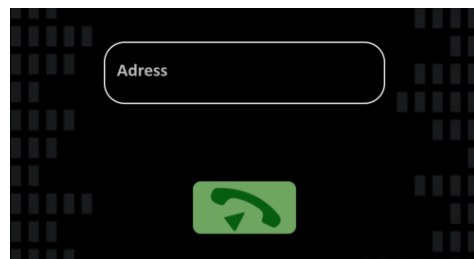


- **DIAL:** This option allows the user to directly dial the destination by typing its address.

Depending on what the current communications mode is (RTP or P/DSIP), the appearance of this screen will change, asking us for the destination's IP address + port, or its URI, respectively:



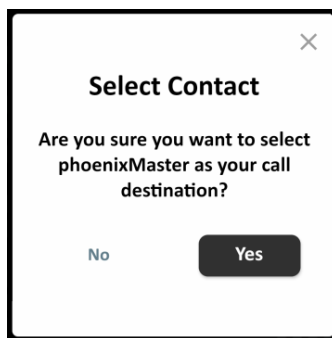
DIAL screen when in RTP mode



DIAL screen when in PSIP or DSIP mode

The text typed in the DIAL screen is validated by touching the green call button under the text box/es.

Once a destination has been selected or typed in by any of the three methods described above, confirmation will be requested and, if acceptance is provided, it will be stored as the next call destination and the App will return to its main screen.



In order to make a call, long-press the green CALL button in the main screen. A pop-up window showing the currently selected destination will appear, asking us for confirmation. If we select Yes, TALENT will initiate a call. If successfully established, the CALL button will become steadily illuminated after a while, as will be the CONNECTED indicator.

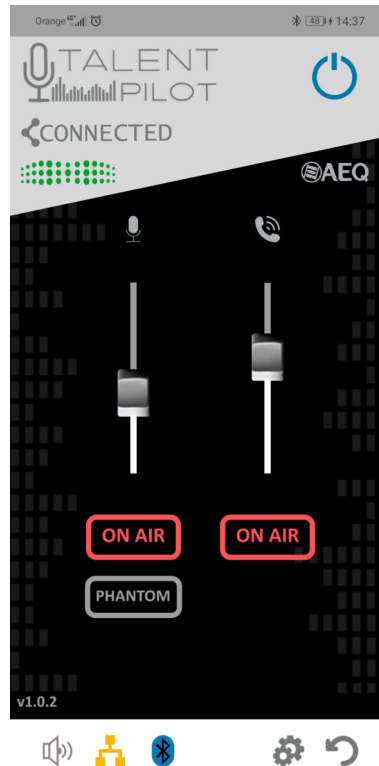
A call can be finished by touching the HANG UP button; confirmation will also be requested (note that if the call is hang up by means of the physical key, it must be pressed for a longer time).

5.5. Audio adjustments.

The central round left and right buttons in the main screen provide access to the audio input and output menus, respectively.

5.5.1. Inputs adjustment menu.

It is possible to modify the microphone gain level (left slider) between 0 and 60dB, as well as sending it to Program (ON AIR button) and activate or deactivate its Phantom power supply, usually required by condenser microphones.



The slider at the right acts on the TELEPHONE output level, not on the auxiliary / Bluetooth input sensitivity level as in "AEQ ControlPHOENIX". This slider is useful to control the level of audio send by the phone, in case it is paired in A2DP and this function is used.

NOTE: If audio is provided through the mini-jack rear input or from a Bluetooth audio link to a different smart device, this fader won't affect the level of that audio signal.

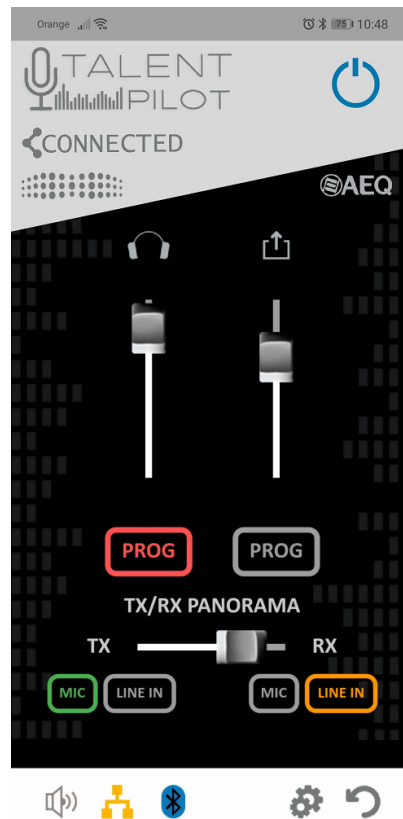
The "ON AIR" button under this slider activates or deactivates sending this auxiliary / Bluetooth input to program. If it is activated, the indicator at the right of the left encoder will become red, just as in the TALENT front panel.

In order to return the main screen, just touch the lower right button:



5.5.2. Outputs adjustment menu.

It is possible to adjust the level of the headphones and auxiliary / Bluetooth output, as well as to select what source/s are listened to at each of these outputs.



The slider button at the left acts on the headphones volume. The buttons located under it ("PROG", "MIC" and "LINE IN") select which sources are heard (Program or local microphone and auxiliary / Bluetooth inputs feeds, respectively). These buttons are not exclusively, so several sources can be mixed together.


The slider button at the right acts on the auxiliary / Bluetooth output. The "PROG", "MIC" and "LINE IN" buttons have exactly the same source-selecting function for this output. If the MIC signal is sent to it, the indicator at the right of the right encoder will illuminate red when returning to the main screen, as will do the physical indicator by the corresponding encoder in TALENT's front panel.

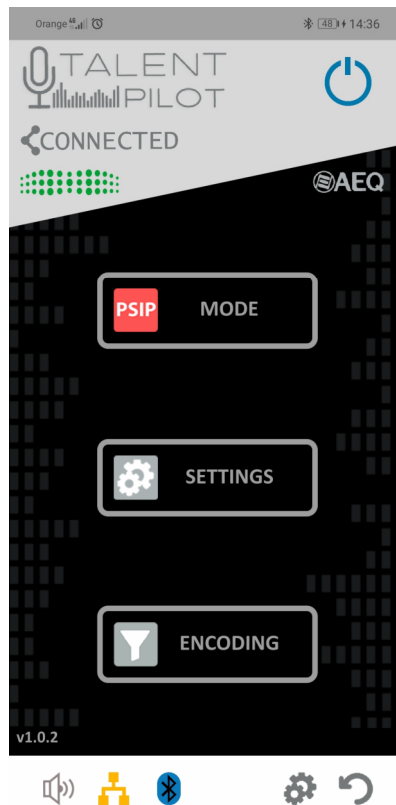
The "TX/RX PANORAMA" slider allows the user to adjust the balance between program local send (TX) and reception (RX) for those outputs that have the corresponding "PROG" button activated.

In order to return the main screen, just touch the lower right button:



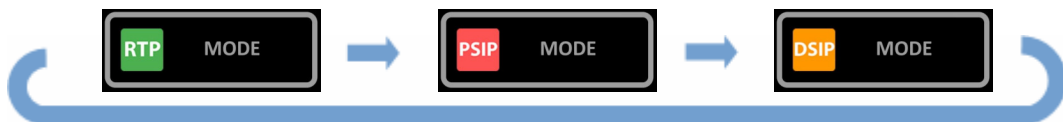
5.6. Settings screen.

This screen is accessed by touching the lower gear icon. 



This screen has three buttons:

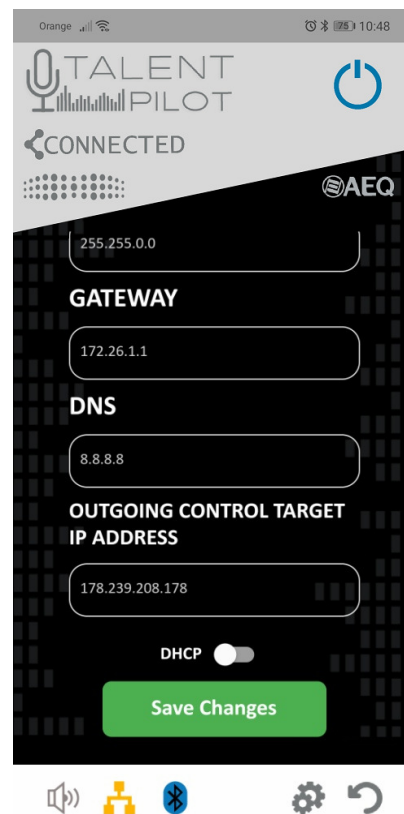
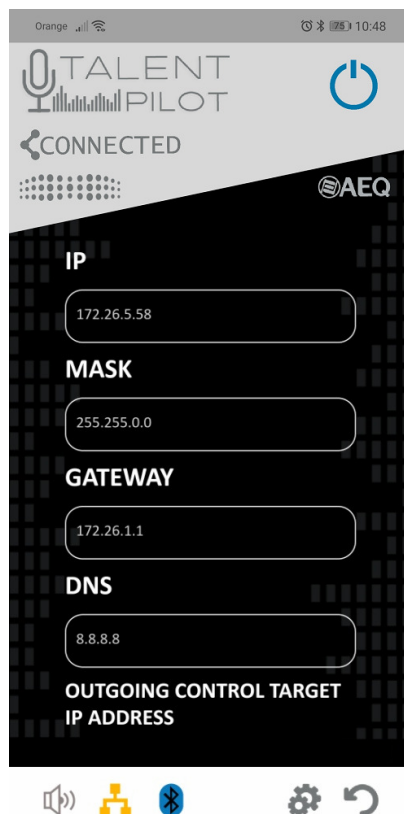
- **MODE:** This button cyclically changes between the different communications modes: Proxy SIP (PSIP), Direct SIP (DSIP) or RTP, each time we touch it:



- **SETTINGS:** This button gives access to different configuration options:



- **IP Config:** Opens a screen where the IP configuration can be checked and/or modified (IP address, subnet mask, Gateway, DNS, DHCP on/off and selection of the outgoing control IP address). It is possible that all that information doesn't fit the screen, so the lower options can be accessed by scrolling up with the finger:



When any of the values is modified, we can apply the changes by touching the "Save Changes" button, that will ask for confirmation:

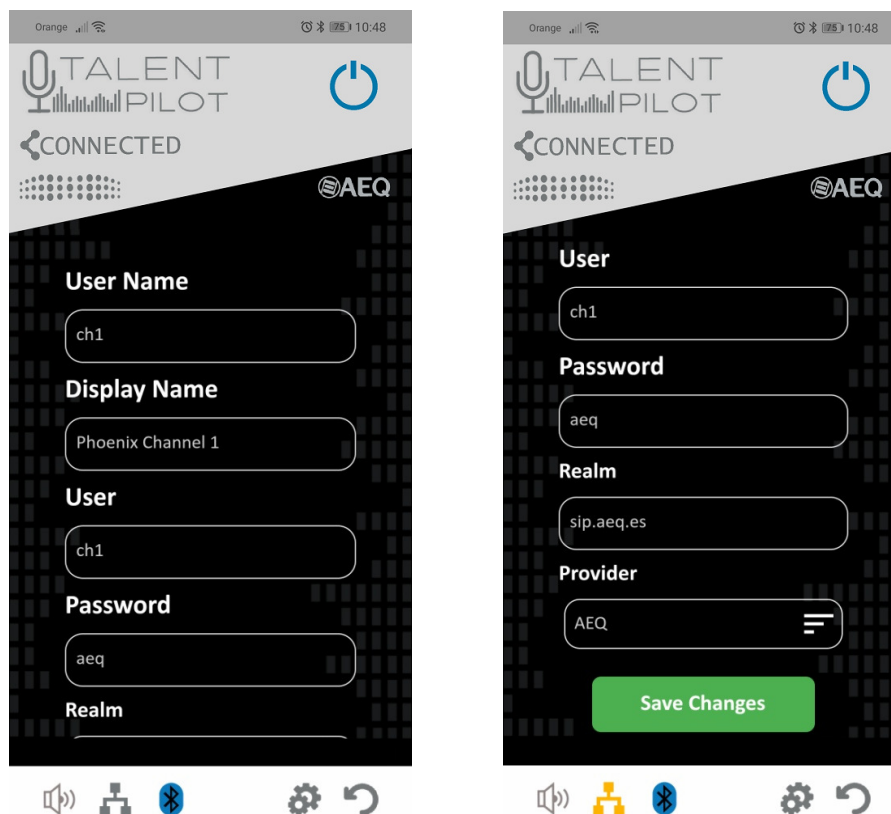


Once acceptance is provided, TALENT will reboot to apply the changes (unless the only modified field is "OUTGOING CONTROL TARGET IP ADDRESS", that doesn't require the unit to be re-booted to be applied).

- **SIP Settings:** Opens a screen where the SIP register parameters can be adjusted.

The parameters to be adjusted are: "User Name", "Display Name", "User" (usually, these three fields contain the same value), "Password" and "Realm" (domain).

Besides, one of the "Providers" (SIP servers pre-stored in TALENT using "AEQ ControlPHOENIX" software) needs to be selected.



- **Reboot:** Re-boots TALENT.

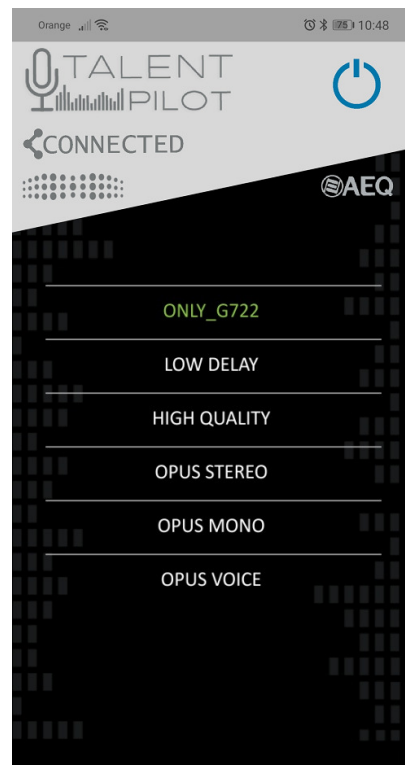
In order to return the main screen, just touch the lower right button:



- ENCODING:** This button opens a list of all the available coding modes. The list looks different when the unit is in RTP mode (it will be a list of all particular coding algorithms) or in any of the two SIP modes (then it will be a list with "profiles" or bundles of coding algorithms previously stored in the unit using "AEQ ControlPHOENIX").




Coding algorithms list (RTP mode)

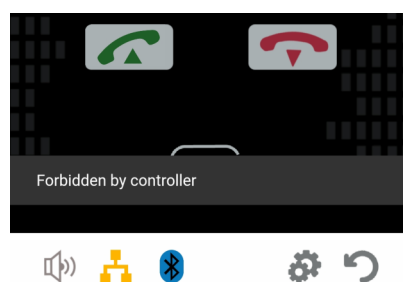


Coding profiles list (SIP mode)

The currently used coding algorithm or profile will be highlighted in green in this list (it may be down the list so, if you don't see it, just scroll with the finger). A new one can be selected by simply touching it.

In order to return the main screen, just touch the lower right button: 

NOTE: "AEQ ControlPHOENIX" remote control software allows the controller personnel to enable or inhibit several local controls (under the Settings / Miscellaneous / User interface locks & options menu), such as making calls, hanging them up, adjusting audio levels, etc. All these permissions will also affect "AEQ TALENT PILOT" App. Other permissions will affect the App only, like changing IP or SIP configuration, communications mode between RTP/PSIP/DSIP or coding algorithms/profiles. If you try to perform from the App any action that is not allowed, changes will make no effect and, additionally, a warning message will appear in the App.



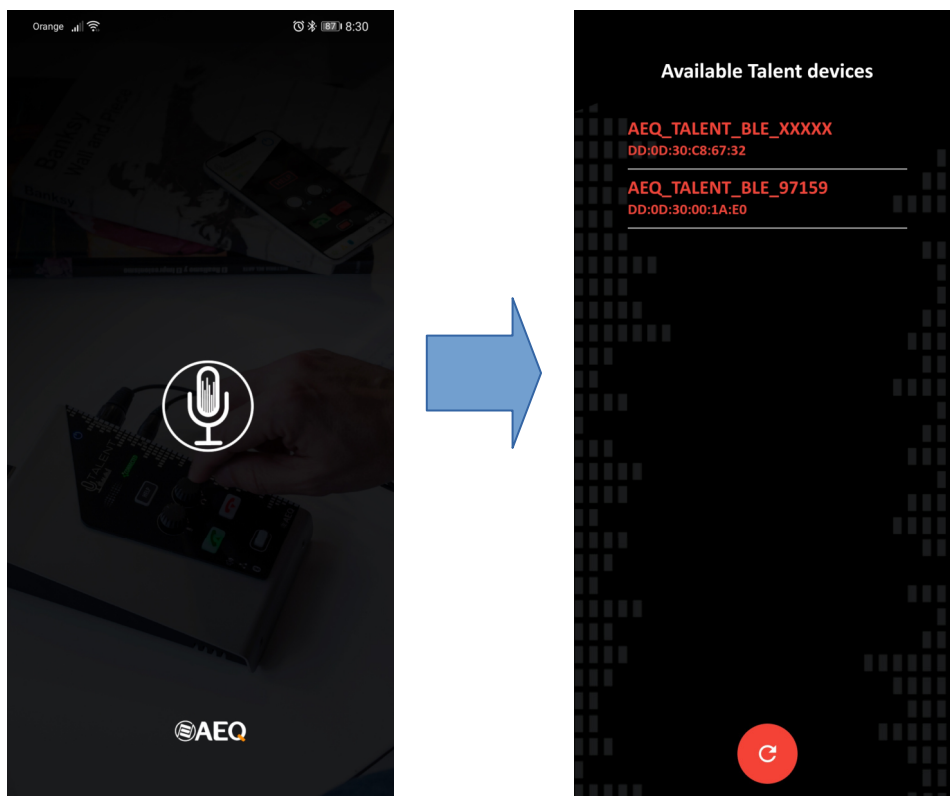
6. QUICK START GUIDE.

6.1. First steps. "AEQ TALENT PILOT" control App.

We strongly recommend you to install this App that provides remote control of the TALENT audiocodec using your smart phone. Depending on whether you are using an Android or iOS terminal, please download "AEQ TALENT Pilot" App from App Store or from Google Play Store. For more information, please check chapter 5 of this manual.

Provide power supply to your TALENT unit using the supplied AC/DC adapter, and press the "POWER SWITCH" button at the back. When the blue Bluetooth LED at the front is blinking, enable Bluetooth and location in your phone, wait for approx. 15 seconds and then run the App "AEQ TALENT PILOT" (providing the requested permissions the first time).

A list of TALENT devices at reach will be displayed after an intro screen. Choose your device according to its serial number, which can be found written at the back of the unit: "AEQ_TALENT_BLE_<serial number>". Touch on its name and the App will connect to it.



From then on, you have basic control of your TALENT from your mobile phone, and you will be able to make calls, configure audio routings and level, operating mode, and some more adjustments, with real-time control of the unit and total coherence with its physical controls.

NOTE: "AEQ ControlPHOENIX" remote control software allows for the inhibition of several local controls (under the Settings / Miscellaneous / User interface locks & options menu), such as making calls, hanging them up, adjusting audio levels, etc. All these permissions will also affect "AEQ TALENT PILOT" App. Other permissions will affect the App only, like changing IP or SIP configuration, communications mode between RTP / PSIP/ DSIP or coding algorithms / profiles. If you try to perform from the App any action that is not allowed, changes will make no effect and, additionally, a warning message will appear in the App.

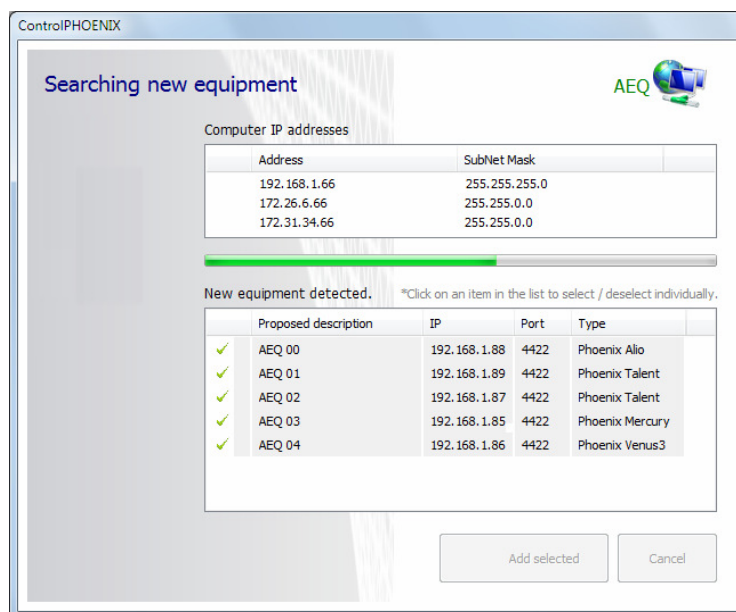
6.2. Connecting to a network.

Connect your TALENT's back Ethernet port to a free LAN port of your home router with access to Internet, or even to a mobile router with internal SIM card. Please refer to section 2.2.4 of this manual.

TALENT is configured from factory to obtain its IP configuration automatically (DHCP). Most home routers have DHCP activated. However, advanced users can assign a fixed IP address from "AEQ TALENT PILOT" App or using "AEQ ControlPHOENIX" configuration software.

6.3. "AEQ ControlPHOENIX" configuration software.

If you have a Windows PC, you can install "AEQ ControlPHOENIX" software, supplied for free. From it, you can have comprehensive remote control of the unit, even through internet. Once installed, please open it and the program will search for AEQ audiocodecs in that PC's configured LAN network/s. In order to know more about "AEQ ControlPHOENIX", please download its manual from the "Manual" link at <http://www.aeq.es/productos/controlphoenix-gestion-de-audiocodecs>



Please note that, if you are a former AEQ codec user, you may need to upgrade your "AEQ ControlPHOENIX" version to 2.2.1.47 or above in order to be able to control TALENT devices.

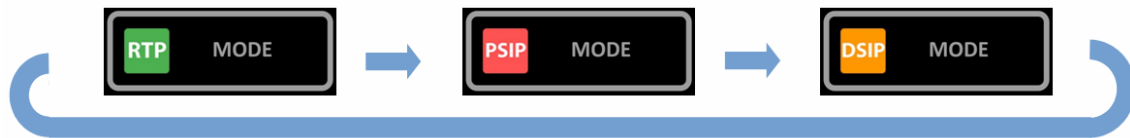
Once your TALENT codec is found, please make sure that it has a green check at its left (otherwise, just click on it) and then click on "Add selected". The software will ask you for a name for that unit within the installation.


From that moment on, you have total control of your TALENT from the PC. We strongly suggest you to read "AEQ ControlPHOENIX" user's manual, available in our website. One of the things that you can do is assigning the unit a fixed IP address, if that's what you need. Press the "HELP" button on the unit and check that a message appears on the PC screen.


6.4. Making your first test call.

You can now make a SIP test call to our test audiocodec.

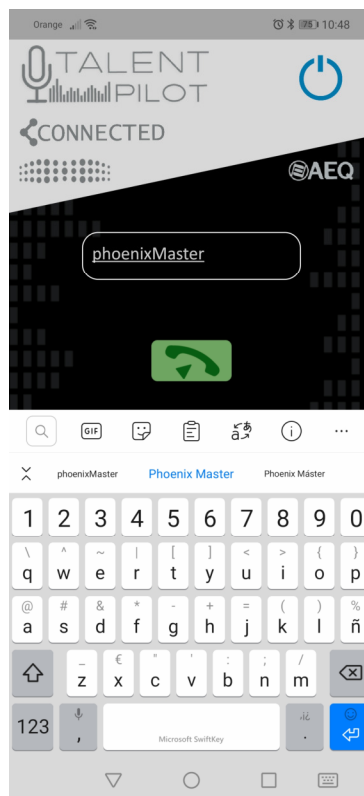
In order to select the Proxy SIP mode, from "AEQ TALENT PILOT" App go to the configuration menu (gear icon at the bottom of the screen) and touch the "MODE" button until it indicates "PSIP".



Then, return to main screen by touching the lower right button. 

Check that the LAN indicator is ON and doesn't blink. That indicates that TALENT is properly registered in AEQ **sip.aeq.es** free server. 

If that's the case, touch the green call button in the App. Among the different call options, select "DIAL" and touch the address field. A keyboard will appear; use it to type the following address: **phoenixMaster**



Next, touch the green button below in order to establish that address as the call destination, confirm and the App will return to the main screen. Once there, **in order to make the call just hold the green CALL button** (in the App or in the TALENT itself) and accept the confirmation (in case you launch the call from the App). The unit will connect to our test unit, which is always returning music.

If you have a microphone and headphones connected, you can act on the audio controls to activate your microphone, adjust its level, listen not only to what's received from the test unit but also to what you are sending to it, the headphones volume, etc.

6.5. Pairing for audio exchange.

If you want to send music to your TALENT from mobile phones or PCs or make calls and send them to the Program, please pair it for audio.

In order to do that, please activate Bluetooth in your phone, and when TALENT is on, search for a device called "AEQ_TALENT_<serial number>" (be aware that you don't have to pair the device "AEQ_TALENT_BLE_<serial number>"), and pair it for Audio and Calls, accepting the confirmation requests that may appear. No password is required.

Once the device is paired for audio, the Bluetooth blue light will be steadily illuminated.



NOTA: It is not mandatory to pair the audio device to be able to control the unit from "AEQ TALENT PILOT" App. In fact, you can pair to a different smart device for audio.

7. CONTROL TERMINAL OVER WEB BROWSER.

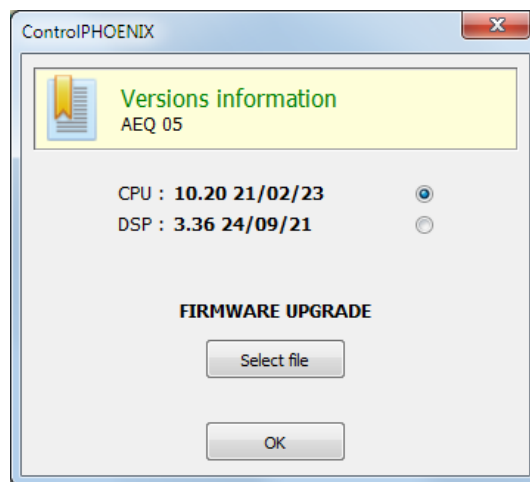
TALENT audiocodec includes a WebServer that enables you to execute numerous functions remotely over the Ethernet interface included in the back panel of the unit by means of a standard web browser (compatibility is guaranteed with Internet Explorer and other browsers running on Microsoft Windows operating system).

7.1. Upgrading system firmware.

TALENT is supplied from factory with the latest firmware versions available. However, firmware versions with new features may be released in the future, making it necessary to upgrade the equipment to be able to make use of these new functionalities.

Because the upgrading process must be handled with caution, we recommend having it done by an authorized distributor, or under the instructions of the AEQ Technical Service. If you have any doubt, please consult AEQ Technical Service (support@aeq.es).

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



IMPORTANT NOTE: If the **CPU** of the equipment is upgraded, configuration of the unit and, in particular, its IP configuration won't be modified unless expressly stated by AEQ SAT. In that case, the user should take note of all important codec parameters before upgrading in order to reconfigure them afterwards.

The entire firmware versions upgrading process is done through the IP interface of the unit, with the aid of a standard web browser (compatibility is guaranteed with Internet Explorer running on Microsoft Windows operating system). By default, user and password is "aeq".

IMPORTANT NOTE: the recommended order for upgrading is: MICRO (CPU), DSP and FPGA. The process is iterative.

To upgrade the firmware, you must follow the steps detailed below:

1. From **MENU** → **ETHERNET**, check the IP address associated with the Ethernet interface.
2. Power down the TALENT.
3. Connect TALENT to the PC from which you are going to perform the upgrading process using a crossed cable.
4. Power up the TALENT.
5. Open the Internet Explorer web browser and, in the address bar, enter HTTP://<IP address obtained in point 1>. Press ENTER and the main screen will be displayed.



Main screen detail

6. To upgrade the codec, click on the **UPGRADE** option.
7. A user ID and password are requested (by default, both are **aeq**). After you have correctly entered these two items, the firmware upgrading screen will be displayed.



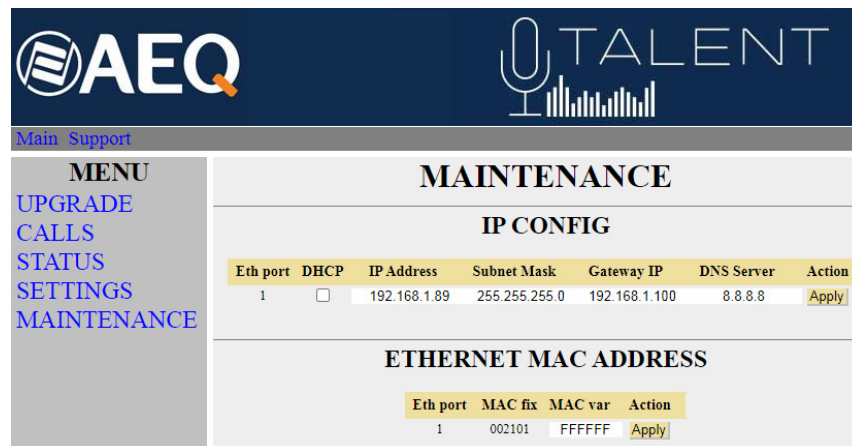
Firmware upgrading screen detail

8. Check to see whether the versions displayed are the same as the firmware that is currently in effect. If they do not match, upgrade the firmware as indicated below.
9. Select the module you want to upgrade in "Upgrade" column. **NOTE:** Each upgrading file is specifically designed to upgrade a specific module within the unit: CPU, DSP or FPGA.
10. In "**Select file**" enter the access route to the upgrade file containing the new firmware version, using the "**Seleccionar archivo**" button.
11. Press the "**Upgrade**" button found in the lower part of the screen.
12. Wait for on-screen confirmation that the operation has been successfully completed.
13. In the Internet browser, go to the **UPGRADE** section and ensure that all the firmware versions installed in your codec are now the correct ones.
14. Power the unit down.

7.2. Configuring the IP parameters and the MAC address associated with the Ethernet interface.

From this menu the IP parameters and the MAC address associated with the Ethernet interface can be edited. Because of the consequences the MAC address configuration could have, this action should only be carried out if the codec use situation requires it; the editing should be performed by highly qualified personnel or under the supervision of AEQ authorized technical services, and always in possession of the required network information.

1. From **CONFIGURATION** → **ETHERNET**, check the IP address associated with the Ethernet interface.
2. Power down the TALENT.
3. Connect TALENT to the PC from which you are going to perform the upgrading process using a crossed cable.
4. Power up the TALENT.
5. Open the Internet Explorer web browser and, in the address bar, enter HTTP://<IP address obtained in point 1>. Press ENTER and the main screen will be displayed. Selecting the **MAINTENANCE** option will enable you to modify the IP parameters and the MAC address of the Ethernet interface of the unit.



Ethernet interface screen detail

6. Modify the value of the IP parameter or the MAC field associated with the Ethernet interface.
7. Press the "**Apply**" button.
8. In the Internet browser, go to the **MAINTENANCE** section and check that the configuration of IP parameters or MAC address is now correct.
9. Power the unit down.

7.3. Technical Service and on-line manuals.

Clicking on the "**Support**" tab in the upper part of the screen will take you to **AEQ** website, where you will find all the information you need to directly contact the AEQ Technical Service, as well as all the technical information and manuals regarding the unit.

7.4. Saving and loading configurations.

In the **MAINTENANCE** section, in the lower part of the screen, you will see the **CONFIGURATION MEMORY** option, from which you can save the current configuration of the unit (by means of "DOWNLOAD" button) or load a configuration previously created and saved (by selecting the corresponding file and pressing then the "Save configuration" button). The extension of the files used in this process is "AFU". The "Reset configuration" button allows you to restore the default configuration of the unit.

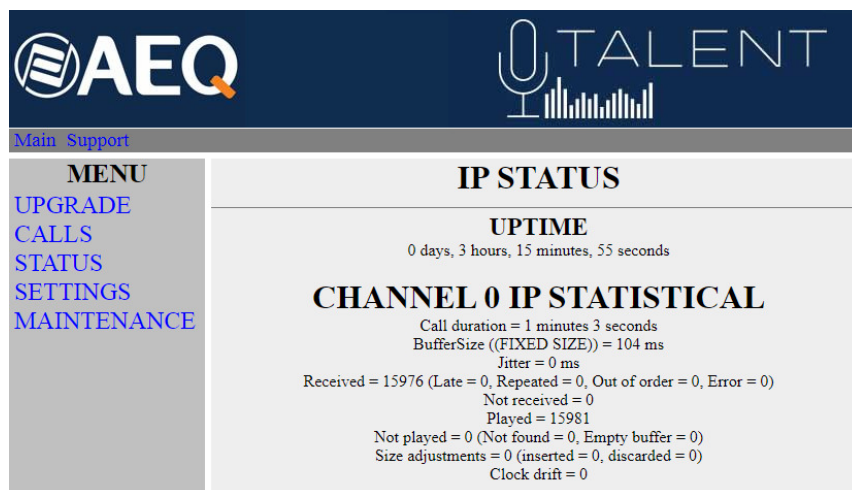


Configuration screen detail

7.5. Status menu.

By means of IP Status menu you can monitor some statistical parameters regarding the connection status of IP channels. Some of these parameters are: transmission and reception buffers status, Jitter, lost packets...

This information can also be accessed with a more elaborated presentation from the **AEQ ControlPHOENIX** application (see section 5.5 of "AEQ ControlPHOENIX" application manual).



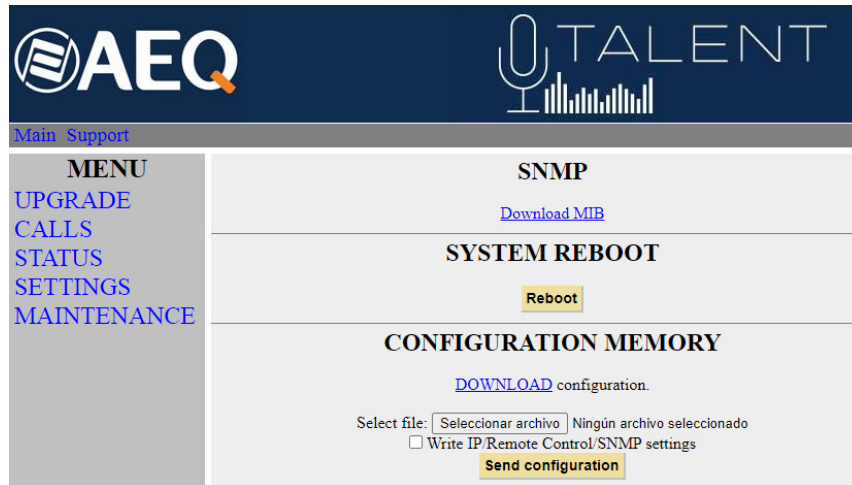
IP Status screen detail

7.6. SNMP.

This unit can be remotely managed using **SNMP** (Simple Network Management Protocol) using one of the many client pieces of software available in the market, even for free. SNMP allows monitoring of the status of several pieces of equipment from very diverse manufacturers and natures, as well as elaborating reports, generate e-mail alerts, etc.

In order to add and equipment to the list of units managed by the client, it is necessary to have access to its "MIB" file (Management Information Base), that describes its SNMP capabilities (alarms it can generate, accepted commands, manufacturer information, etc.).

The MIB file corresponding to the unit can be downloaded from the Web interface without installation of any additional software. In order to do so, in the MAINTENANCE section, you can access the link "**Download MIB**" under the "**SNMP**" section.



SNMP screen detail

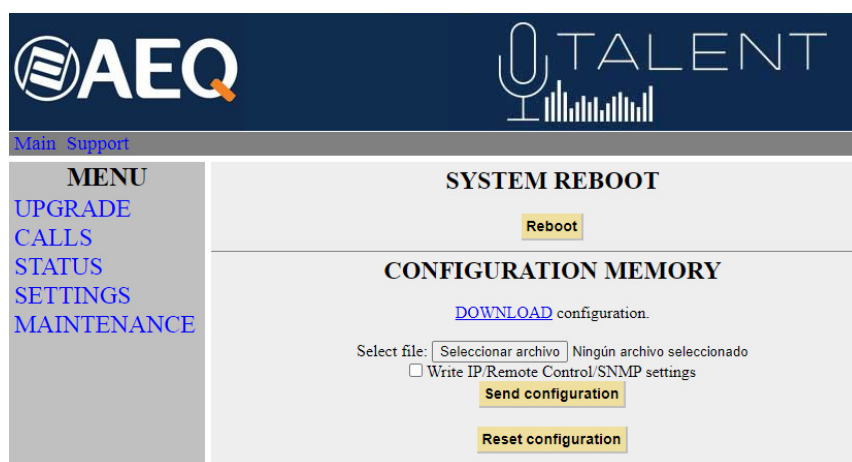
If you follow that link, the text file will appear. Now you just need to right-click on it and select "Save as..." and browse a suitable destination folder (see the manual of the selected SNMP client).

For more information, please consult section 4.7 of this manual and section 6.5.1 of "AEQ ControlPHOENIX" application manual.

7.7. Remotely rebooting the equipment.

Sometimes a remote unit can work abnormally or become unreachable by the remote control software as a result of a communications error or in the unit itself. A method has been developed in order to remotely reboot the audiocodec, so normal operation is recovered.

Inside MAINTENANCE, at the bottom of the screen, you can find "**SYSTEM REBOOT**" section. By clicking on "**Reboot**" button, an information dialog will appear warning that the equipment is being rebooted and it will be disconnect for some seconds after acceptance ("Resetting system... please wait a few seconds before connecting again").



Reboot screen detail

8. TECHNICAL SPECIFICATIONS *

Audio inputs and outputs	
Dynamic or condenser MIC inputs	Female XLR-3 connector. Input impedance: 2 K Ω . Valid for most microphones with nominal 48V Phantom. High quality, low noise, electronically balanced preamplifier
Stereo analogue line input	3.5mm female jack. Input impedance > 15 K Ω .
Bluetooth audio input	A2DP profile for high-quality audio reception (Bluetooth 5.0 with AAC encoding, among others). HFP voice profile (with G711 or G722 encoding, depending on the App used).
Stereo analogue line output	3.5mm female jack. Output impedance: < 50 Ω
Headphone output:	1/4" stereo female jack. For low or high impedance headphones, between 16 and 500 Ω .
Bluetooth output	A2DP profile for high-quality audio reception (Bluetooth 5.0 with AAC encoding, among others). HFP voice profile (with G711 or G722 encoding, depending on the App used).
Audio characteristics	
Max output line level	835 mV. RMS (-1.5dBV), THD <0.5%
Max input line level	1 V RMS (0dBV)
Distortion	< 0.011 % on the way MIC→Line Out or MIC→HP (Vin=-35dBu, Gin=33dB, Gout=0dB, BW=22kHz)
Dynamic range	> 87 dB on the way MIC→Line Out or MIC→HP (Vin=-35dBu, Gin=33dB, Gout=0dB, BW=22kHz)
Binary rate	12 to 2304 Kbps
Operating modes	Mono, Stereo, Dual, JStereo, MS Stereo
Frequency response	+/-0.5 dB @ 20 Hz-20 KHz
A/D & D/A conversion	
Analogue I/O: A/D & D/A converter	24 bit Sigma-Delta 48 KHz
Communications interfaces	
IP interface	Ethernet RJ45 10/100Mbps port
Coding algorithms	OPUS, G.711, G.722, MPEG Layer 2, PCM... (See APPENDIX A)
Control and data interface	
LAN 10 / 100 base T	RJ45 connector
General characteristics	
Power supply	5 to 15 VDC for external adapter (90-263 V 50/60 Hz input) or 5V through USB interface. Power consumption: 3.5 W. max.
Operating temperature	-10 to +45 ° C (14 to 114 ° F)
Weight	620 grs.
Dimensions	Width x Depth x Height: 130x195x45 mm.

* Characteristics are subject to changes without previous notice.

Safety regulations

CE Marking.

Electromagnetic compatibility according to EU directives EN 50081-1, EN 50052-2.

9. A.E.Q. WARRANTY.

AEQ warrants that this product has been designed and manufactured under a certified 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 warranty does not exclude or limit in any way any legally recognized right of the client.

2.- The warranty period 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 warranty, 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 warranty period, 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 warranty.

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 warranty does not include the freight charges of the product to or from such Authorized Technical Service Centre.

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

5.- The present warranty 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 Warranty 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.

APPENDIX A: General characteristics of encoding modes.

OPUS:

OPUS is a completely open and very versatile coding algorithm. Its performance is unrivalled for voice and audio transmission. It was standardized by Internet Engine Engineering Task Force (IETF) as [RFC 6716](#), and combines Skype's SILK codec technology with Xiph.Org's CELT.

This algorithm allows for an excellent audio quality with high compression rate and very low delay. Phoenix family audiocodecs feature 7 selected OPUS modes covering nearly every transmission need, from voice to high quality stereo music, with bitrates between 12 and 192 Kbps and audio bandwidth between 6 and 20 KHz. The receiver can automatically adapt to the particular OPUS mode selected in the transmitting end.

G.711:

ITU encoding standard for processing audio signals in the human voice frequency band, through the compression of digital audio samples obtained at 8 KHz, and typically used in telephone systems.

Bandwidth: 3.5 KHz

For further information on this subject, consult:

<http://www.itu.int/rec/T-REC-G.711/e>

G.722:

ITU encoding standard, based on ADPCM algorithms, for processing audio signals in the human voice frequency band, through the compression of digital audio samples obtained at 16 KHz, for greater audio quality and clarity.

This is the internationally accepted mode for two-way communication because of its low delay, which is why it is the most used standard in commentator and sports broadcasting applications.

Bandwidth: 7 KHz.

For further information on this subject, consult:

<http://www.itu.int/rec/T-REC-G.722/e>

MPEG LAYER II:

Well known, widely accepted encoding mode that is used when the delay is not important, since MPEG modes always have a greater delay than G.722 modes. There are 64 Kbps encoding modes with sampling rates of 48, 32 or 24 KHz, and 128 Kbps encoding modes with sampling rates of 32 and 48 KHz.

Bandwidth: 10 KHz to 15 KHz.

For further information on this subject, consult: ISO/IEC 11172-3 and ISO/IEC 13818-3.

AEQ LD+:

AEQ proprietary mode (based on the previous AEQ-LD Extend mode) that combines the low delay offered by G.722 with the greater bandwidth of the MPEG modes, optimizing these two aspects.

PCM (12/16/20/24 bits):

Linear audio without any compression process.

For further information on this subject, consult

<http://www.digitalpreservation.gov/formats/fdd/fdd000016.shtml>

Other different encoding modes can be taken into account according to specific needs of each client.

Available coding algorithms list in TALENT. Please contact us to check availability of other algorithms.

CODEC	RATE (Kbps)	Fs (KHz)	Mode	Approx audio BW (KHz)
OPUS	12	48	VOICE RB	6
	20	48	VOICE	8
	32	48	MUSIC MONO RB	20
	64	48	MUSIC MONO	20
	64	48	MUSIC STEREO RB	20
	128	48	MUSIC STEREO	20
	192	48	MUSIC STEREO HQ	20
G.711 A-law	64	8	MONO	4
G.711 μ -law	64	8	MONO	4
G.722	64	16	MONO	7.5
AEQ-LD	64	16	MONO	7.5
	128	32	MONO	15
	192	48	MONO	19
	128	16	STEREO	7.5
	256	32	STEREO	15
	384	48	STEREO	19
MPEG-1 Layer II	64	32 / 48	MONO	10.5
	128	32 / 48	M / ST / JST	10.5/16.5
	192	32 / 48	M / ST / JST / D	10.5/16.5
	256	32 / 48	ST	16.5
	384	48	STEREO	16.5
MPEG-2 Layer II	64	16 / 24	MONO	8/12
	128	24	MONO	12
PCM (12,16,20,24 bit/sample)	384 a 768	32	MONO	16
	768 a 1536	32	STEREO	16
	576 a 1152	48	MONO	20
	1152 a 2304	48	STEREO	20

Other different encoding modes can be taken into account according to specific needs of each client.

APPENDIX B: Protocols associated with IP communications.

Communication over IP networks differs notably from the communications traditionally used to date in broadcast environments, whether they are POTS or ISDN, in that IP networks do not have dedicated resources or qualities of service implemented in most systems, with the associated problems this involves in terms of communication signalling, establishment, maintenance and clear down.

This set of problems originates in the technical characteristics that are intrinsic to the definition and operation of communications systems based on IP protocols. The EBU-TECH 3326 standard developed by the N/ACIP working group provides certain tools for attempting to simplify work by making use of many protocols associated with IP communication, and which will be described below.

N/ACIP

- Signalling, understood as connection initiating and ending procedures, as well as negotiation of connection parameters (encoding algorithms, ports, etc.)
 - SDP (Session Description Protocol) to describe the parameters of the connection
 - SAP (Session Announcement Protocol) for multicast type unidirectional links
 - SIP (Session Initiation Protocol) simulates the working system in traditional telephone networks
- Transport: defines the transport protocols over IP networks
 - RTP (Real-Time Transport Protocol) over UDP and IPv4
 - RTCP (Real-Time Control Transport Protocol) for synchronization and active retrieval functions
 - IP ports defined: 5004 / 5008 (RTP for PROG / COORD, respectively) and 5005 / 5009 (RTCP)

While this appendix is not intended to be a reference document for all the relevant technical matters, it should at least serve to give its readers an initial contact with these subjects that will ease the assimilation of the new working method over IP networks for the Phoenix ALIO user and, as a result, the use of this equipment. Users interested in expanding his or her knowledge of some or all of these subjects are encouraged to turn to the extensive, excellent technical material currently available regarding the IP realm and the technologies associated with it.

B1. Circuit switching versus packet switching.

The communications systems traditionally used in the broadcast environment for applications with portable codecs have been mostly telephone or ISDN networks—that is, circuit switching networks; Phoenix ALIO, on the other hand, uses a packet switching network in its IP interface.

B1.1. Circuit switching.

In a circuit switching network, the switching equipment must establish a physical path between the communication media prior to the connection between users. This path remains active during the communication between the users, and is cleared down or released when the communication ends. Example: Switched telephone network.

Its operation passes through the following stages: request, establishment, file transfer and connection clear down.

B1.1.1. Advantages:

- The transmission is made in real time.
- Dedicated resources. The nodes that are involved in the communication use the established circuit exclusively as long as the session lasts.

- Once the circuit has been established, the parties can communicate with each other at the highest speed that the medium allows, without having to share neither the bandwidth nor the use time.
- The circuit is fixed. Because a physical circuit is specifically dedicated to the communication session in question, once the circuit is established there are no losses of time for calculation and decision-making regarding routing through the intermediate nodes. Each intermediate node has a single route for the incoming and outgoing packets that belong to a specific session, which means it is impossible for the packets to be disordered.
- Simplicity in the management of intermediate nodes. Once the physical circuit has been established, no further decisions need to be made to route the data from origin to destination.

B1.1.2. Disadvantages:

- Delay in initiating communication. A time interval is required to make the connection, which entails a delay in the transmission of the information.
- Blockage of resources. No use is made of the circuit during the moments when there is no transmission between the parties. Bandwidth is wasted while the parties are not communicating with each other.
- The circuit is fixed. The communication route is not readjusted; it is not adapted at each opportunity to the least costly path between the nodes. Once the circuit has been established, no use is made of the alternative, less expensive pathways that may become available during the session.
- Poor fault tolerance. If an intermediate node fails, the entire circuit crashes. The connections then have to be re-established from zero.

B1.2. Packet switching.

The sender divides the message to be sent into an arbitrary number of packets of the same size, to which a header and the originating and destination addresses are added, as well as control data that will then be transmitted through different communication media between temporary nodes until they reach their destination. This switching method is the one that is used in today's IP networks. It has emerged to optimize transmission capacity through existing lines.

The temporary nodes store the packets in queues in their memories, which need not be very large.

B1.2.1. Switching modes.

- Virtual circuit: Each packet is routed through the same virtual circuit as the preceding ones. Therefore the order of arrival of the packets to their destination is controlled and ensured.
- Datagram: Each packet is routed independently from the rest. Thus the network cannot control the path followed by the packets, nor ensure the order in which they reach their destination.

B1.2.2. Advantages:

- In case of error in a packet, only that packet will be resent, without affecting other packets that arrived without errors.
- Interactive communication. Limiting the maximum packet size ensures that no user can monopolize a transmission line for very long (microseconds), which means that packet switching networks can handle interactive traffic.
- Packet switching increases network flexibility and profitability.
- The pathway a communication takes can be altered from one moment to the next (for example, in case one or more of the routers breaks down).

- Theoretically, priorities can be assigned to the packets in a given communication. Thus, a node can select, from its queue of packets waiting to be transmitted, the ones that have higher priority.

B1.2.3. Disadvantages:

- Greater complexity of the intermediate switching devices, which need to have higher speed and greater calculating capacity to determine the appropriate route for each packet.
- Packet duplication. If a packet takes too long to reach its destination, the receiving device may conclude that it has been lost, in which case it will send a packet retransmission request to the sender, which gives rise to the arrival of duplicate packets.
- If the routing calculations account for an appreciable percentage of the transmission time, the channel throughput (useful information / transmitted information) decreases.
- Variations in the mean transit delay of a packet in the network. Parameter known as jitter.

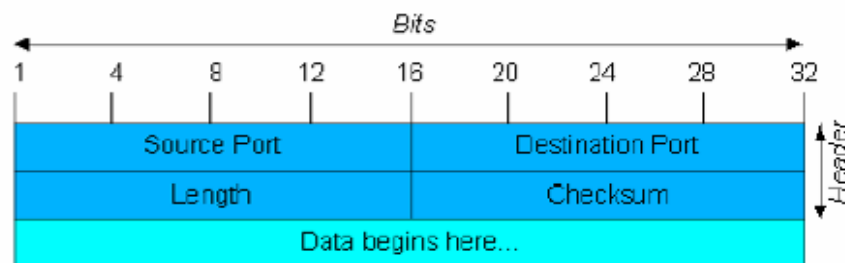
B2. IP protocol.

The Internet Protocol (IP) is a non-connection oriented protocol used both by the origin and the destination in data transmission over a switched packet network.

The data in an IP-based network are sent in blocks known as packets or datagrams (in the IP protocol these terms are used interchangeably). In particular, in IP there is no need for configuration before a device attempts to send packets to another with which it has not communicated previously.

The Internet Protocol provides an unreliable datagram service called UDP (User Datagram Protocol), also known as "best effort", a phrase that expresses good intentions but offers few guarantees. IP does not offer any mechanism to determine whether a packet reaches its destination, and only provides security (by means of checksums) to cover its headers, and not the transmitted data. For example, since it gives no guarantee that the packet will reach its destination, it could arrive damaged, in the wrong order with respect to other packets, duplicated, or simply not arrive. If reliability is needed, it is provided by transport layer protocols such as TCP (Transport Control Protocol).

Reliability over TCP is obtained through the use of retransmissions. Real-time applications such as an audio link, with the timing requirements inherent in the information contained in the link, do not offer any useful guarantee. Since the data that are not received, and whose retransmission is requested of the sender by the receiver, will in most cases arrive out of order, they will end up as useless information that will have served only to overload the network. For all these reasons, the protocol selected to serve as a communication substrate in real-time applications is UDP.



UDP Datagram

Transport over IP protocols, independently of the reliability they offer, add new functionalities to the basic ones provided by IP, such as packet numbering to facilitate, on the receiving end, the detection of losses (although not their correction) and of disorder in the information received; and the advent of the port concept as an identifier of different logic connections over the same IP interface.

For complete information on IP protocol, we recommend consulting:

<http://tools.ietf.org/html/rfc791>

<http://www.iana.org/assignments/port-numbers>

B2.1. IP addressing.

An **IP address** is a number that logically and hierarchically identifies an interface of a device in a network that uses the IP protocol. The format used is X.X.X.X, where each X represents a group of eight bits translated into decimal form—that is, whose minimum value is 0.0.0.0 and whose maximum value is 255.255.255.255.

IP addresses are classified in two major groups: static and dynamic.

- It is typical for a user to connect to the Internet from his or her home using an IP address. This address may change when the user reconnects, and this manner of assigning IP addresses is called a *dynamic IP address* (normally abbreviated as *dynamic IP*).
- The Internet sites that, by nature, need to be continuously connected generally have a *static IP address* (as with the dynamic address, a similar abbreviated form is used: *static* or *fixed IP*)—that is, an address that does not change over time.

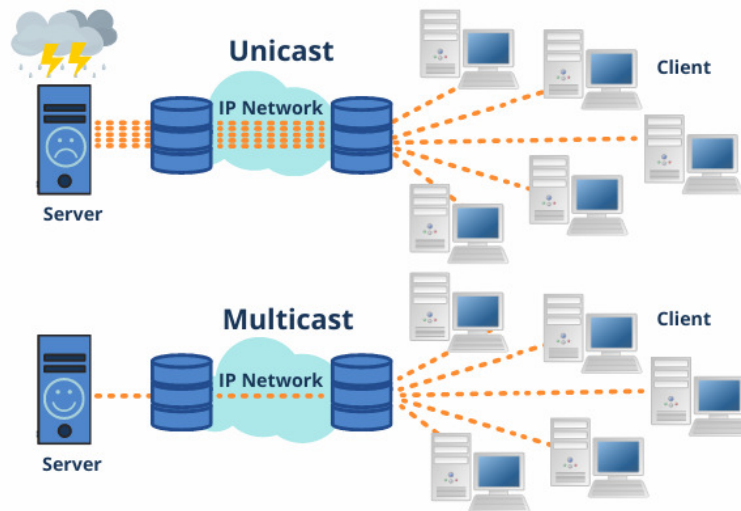
Another possible IP address classification can be made according to address validity:

- **Public:** IP addresses that are valid in the entire Internet network. Currently, due to the poor management that has traditionally been applied to the available IP addresses, they are a scarce, highly costly resource.
- **Private:** addresses that are only valid in a closed section of the IP network, typically corporate and not subject to free access, with only one point of connection to the Internet, called a gateway, constituted by a router.

B2.2. Unicast vs. Multicast.

Unicast is the transmission of information from a single sender to a single receiver. It is distinguished from multicast (transmission to certain specific recipients—more than one—from a single sender), broadcast (in which the recipients are all the stations in the network) and anycast (transmission to a single recipient—any unspecified recipient).

The unicast method is the one currently being used on the Internet, and is applied for both live and on-demand transmissions. The multicast method can only be used in corporate environments, despite some isolated efforts to introduce it on the Internet, and is applied only for live transmissions.



Graphical comparison: Unicast vs. Multicast

The effect that unicast transmission has on network resources is accumulative consumption. Each user who connects to a multimedia transmission consumes as many kilobits per second as the content encoding will permit.

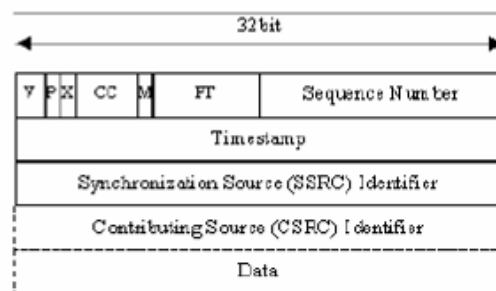
B3. RTP protocol.

RTP are the initials of **Real-time Transport Protocol**. It is a transport level protocol used for the transmission of information in real time, as occurs with audio and video. Normally it is paired with RTCP (RTP Control Protocol) and is located on UDP.

The IP ports defined for its use are 5004 (RTP) and 5005 (RTCP) for PROG and 5008/5009 for COORD.

The functions of the RTP/RTCP protocol are:

- Management of the reception buffer in order to minimize the jitter effect introduced by the network.
- Recovery of the reference clock based on information inserted by the transmitting equipment.
- Test tools to permit the user to verify the bandwidth, the delay and estimated jitter in the connection.



RTP Header

For complete information on RTP/RTCP protocol, we recommend consulting:

- <http://tools.ietf.org/html/rfc1889>
- <http://tools.ietf.org/html/rfc1890>
- <http://tools.ietf.org/html/rfc3550>
- <http://tools.ietf.org/html/rfc3551>
- <http://tools.ietf.org/html/rfc3711>

B3.1. Default TALENT configuration.

TALENT is an IP audiocodec that operates by using RTP over UDP in IP version 4.

By default, TALENT is supplied from the factory with the following IP ports defined: 5004 for RTP and the next one (5005 in this case) for RTCP.

The RTP port values can be modified from its internal menu, and RTCP ports will be automatically assigned accordingly.

B4. SIP protocol.

Session Initiation Protocol (SIP) is a protocol developed by the IETF MMUSIC Working Group with the intention of establishing the standard for initiating, modifying and ending interactive user sessions involving multimedia elements such as video, voice and instant messaging.

SIP is used simply to initiate and terminate voice and video calls. Once the communication is established, the exchange of voice / video information is conducted only over RTP.

One of the objectives of SIP was to contribute a set of processing functions to apply to calls and capacities present in the public switched telephone network. Thus, it implemented typical functions that a common telephone terminal offers, such as: calling a number, making a telephone ring when called, hearing a dial tone or busy tone. The implementation and terminology in SIP are different.

SIP requires proxy servers and register elements to give a practical service. Although two SIP terminals can communicate with each other without the mediation of SIP infrastructures through the use of URIs of the name@IP-address type (which is why SIP is defined as a point-to-point protocol), this approach is impracticable for a public service because of the problems inherent in IP addressing, where obtaining static public addresses is nearly impossible and extremely costly.

To simplify the operation of the unit, AEQ offers, at no additional cost, the services of its 2 own SIP servers (one of them working as main server and the other one as backup server), although it cannot guarantee its operation 100% of the time, nor be held responsible for the inconveniences that this may produce for the end user. The unit leaves the factory preconfigured with the parameters required to work with the resources of any of these 2 SIP servers.

SIP makes use of elements called **proxy servers** to help route the requests toward the user's current location, authenticate users to give them service, enable call routing policies to be implemented, and contribute added capabilities to the user.

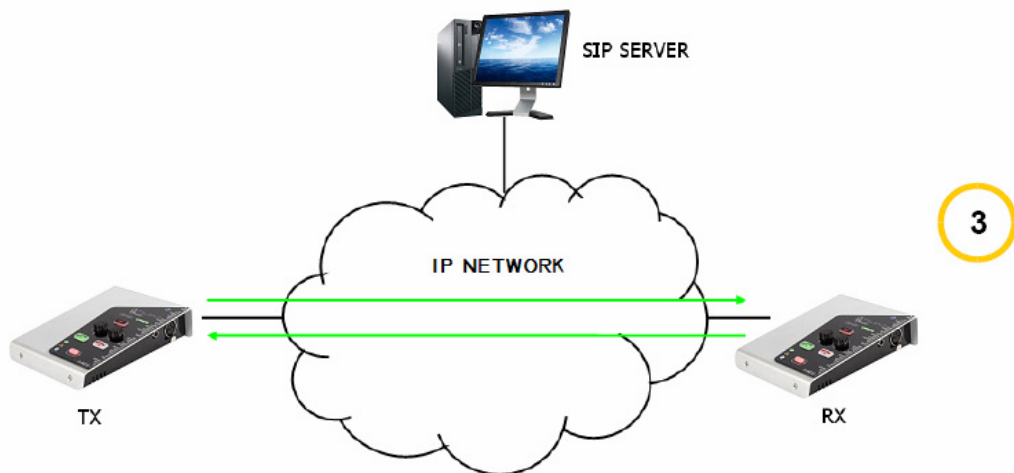
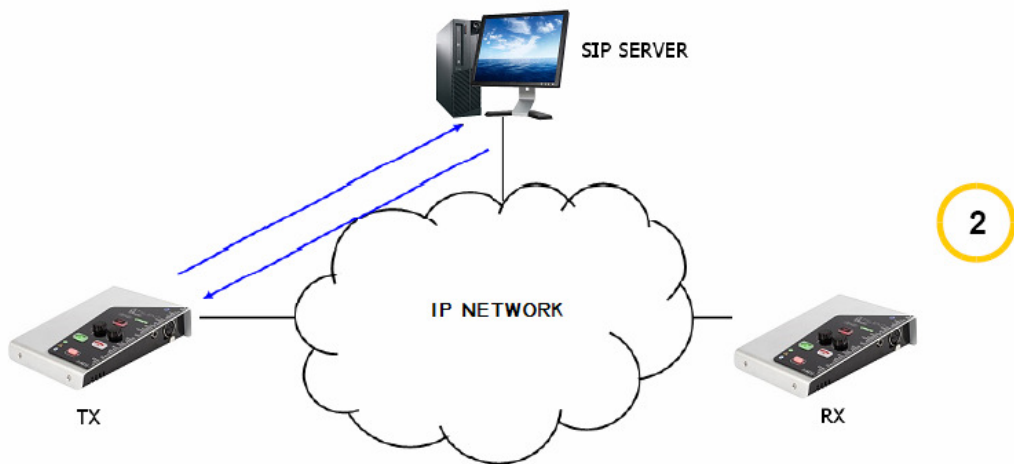
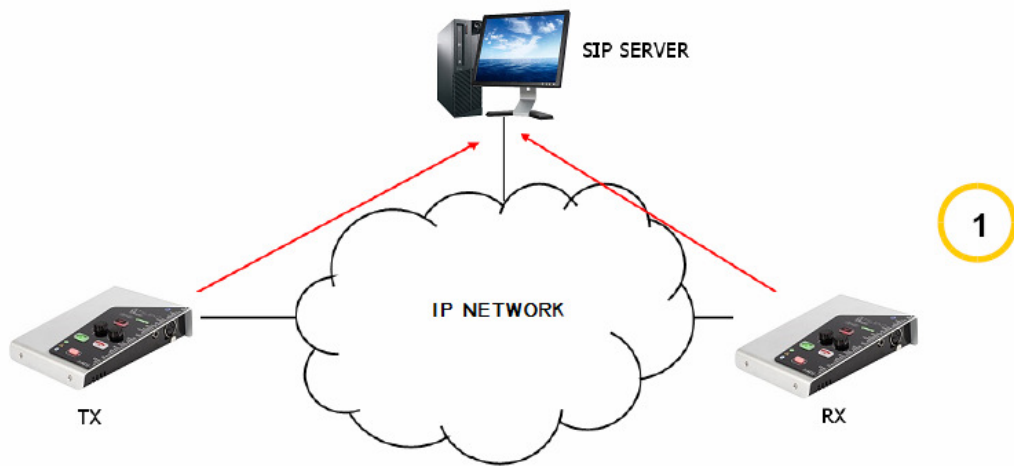
SIP also contributes register functions that enable the user to inform the proxy servers of his or her location.

For complete information on the SIP protocol, we recommend consulting:
<http://tools.ietf.org/html/rfc3261>

B4.1. Working modes.

With the PROXY SIP option activated in the TALENT, when the unit is started up it will automatically connect and register itself in the SIP Proxy server configured in its memory, indicating its name (URI: name@domain) and position (IP address).

To establish any communication, the unit that wishes to establish the connection will search the SIP Proxy server register for the information regarding the called device and will redirect the call—in a way that is transparent to the user—toward the real physical place where the device is located.



SIP protocol operation diagram. Phase 1: Registration. Phase 2: Search for the called device in the SIP server database. Phase 3: Establishment of the connection

This working method, supported by external SIP servers, enables the physical position of a device to be made independent from its logic identifier and, through the use of the SIP protocol, makes it unnecessary to know more data regarding the called device than its URI.

During the establishment of the communication phase, the encoding algorithm is negotiated simultaneously, based on the Link Profiles / SIP Codec Profiles defined in each of the devices at the two ends of the connection circuit.

B4.2. Possible work scenarios.

Depending on the kind of network TALENT is connected to, and leaving apart the trivial case where both involved devices are in the same local network, each unit will present a certain kind of IP address.

- Static public IP addresses offer the ideal situation, as they guarantee that the IP interface of the codec will always be assigned to a fixed address (regardless of whether it is turned off and then powered up again) and directly accessible to the rest of the network users. Phoenix ALIO operates perfectly with an associated SIP server and equipment identifiers of the name@domain type (PROXY SIP), and even without an associated SIP server with a URI of the name@IP-address type (DIRECT SIP) if the device on the opposite end of the communication circuit also has an IP address of the same type.

This situation corresponds to use an Internet access by means of a single workstation router (just one piece of equipment connected) and to hire a fixed IP.

- Dynamic public IP addresses, corresponding to use an Internet access by means of a single workstation router and a dynamic IP (the most usual). Allows the use of URIs of the name@domain (PROXY SIP) or name@IP-address (DIRECT SIP) type, but it is advisable always to work with an associated SIP server (PROXY SIP), since the IP address assigned to the equipment may change each time the user powers up the unit.
- Private IP addresses, both static and dynamic, corresponding to connections in a local network with several workstations, that access to the Internet through a router with NAT. Those do not allow the use of URIs of the name@IP-address type because the IP address of the identifier is not public, and is only valid in the section of the network to which it has been assigned; it lacks a universal meaning. In this case the use of an associated SIP server and a STUN server is imperative to get past the NAT (Network Address Translation) implemented in the router that acts as an interface between the private network and the public one. See section NAT TRAVERSAL (4.3).

B4.3. TALENT default SIP configuration.

To simplify operating the unit, AEQ offers, at no additional cost, the services of 2 own SIP servers.

TALENT is supplied from the factory with both SIP servers preconfigured (SYSTEM → SIP PROVIDERS menu) defined as "AEQ" and "AEQ 2" with the following configuration:

- PROXY SIP "AEQ" Host: sip.aeq.es
- PROXY SIP "AEQ 2" Host: sip2.aeq.es
- PROXY SIP "AEQ" and "AEQ 2" Port: 5060
- PROXY SIP "AEQ" and "AEQ 2" Domain: sip.aeq.es
- PROXY SIP "AEQ" and "AEQ 2" Register Expires: 60 min.

TALENT is supplied preconfigured with **2 users** registered in both servers:

- PROXY SIP "AEQ" and "AEQ 2" Authentication Users: phxalio_XXX y phxalio_XXY (where $Y=X+1$)
- PROXY SIP "AEQ" and "AEQ 2" Authentication Passwords: 8 digits alphanumeric passwords

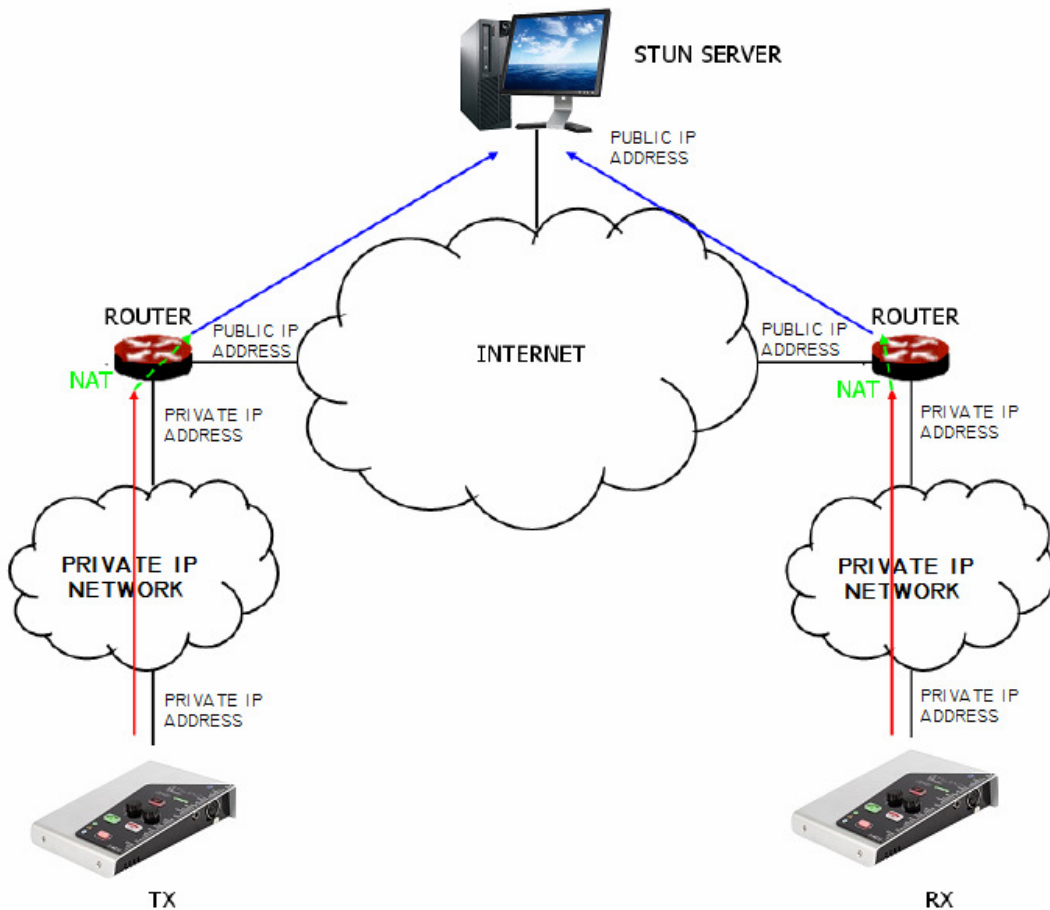
This configuration is the right one for working with any of both AEQ's SIP servers.

B5. STUN protocol.

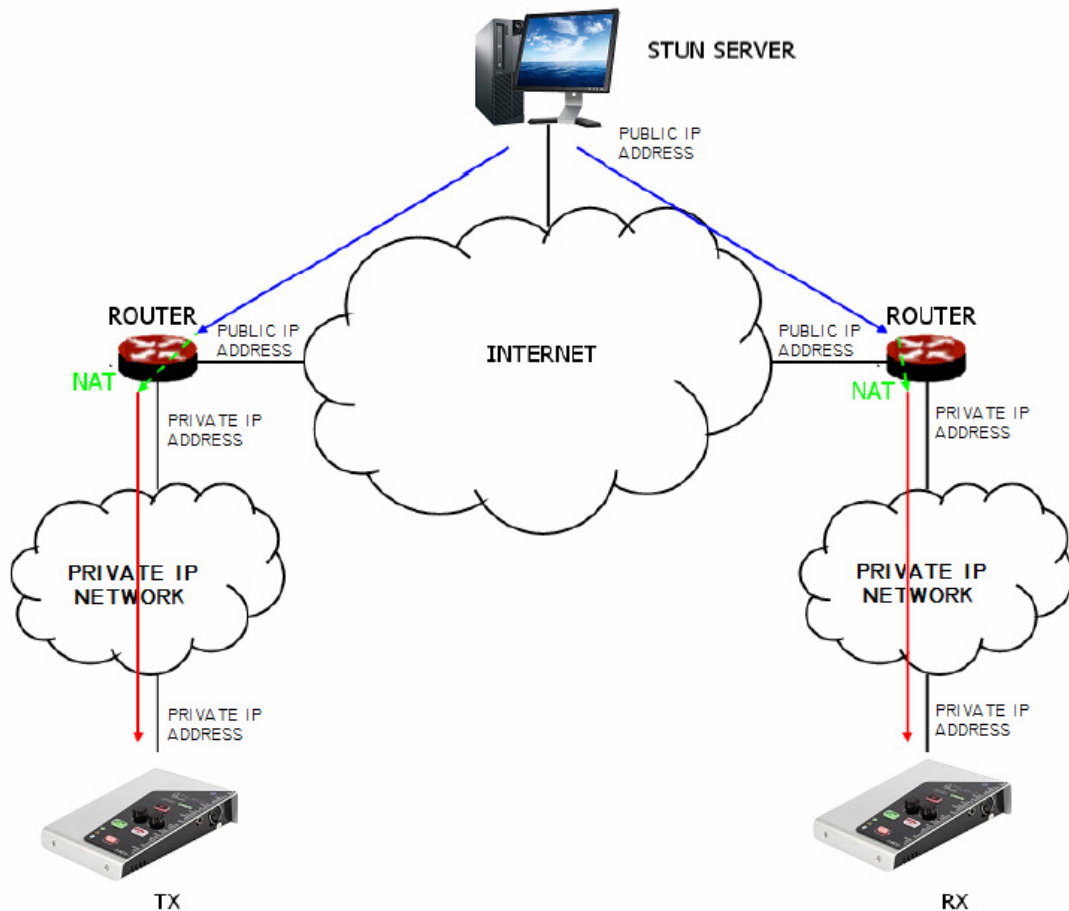
STUN (Simple Transversal of UDP over NATs) is a network protocol of the client /server type that allows NAT clients to find their public IP address, the type of NAT where it is located and the Internet port associated with the local port through NAT. This information is used to configure a UDP communication between two hosts located behind NAT routers.

NAT (Network Address Translation) is a mechanism used by IP routers to exchange packets between two networks that assign each other incompatible addresses. It consists of converting, in real time, the addresses used in the transported packets. It is also necessary to edit the packets to enable the operation of protocols that include address information within the protocol conversation. It is most commonly utilized to enable the use of private addresses and still provide connectivity with the rest of the Internet.

TALENT includes a STUN client that sends a request to a STUN server. The STUN server then informs the client of its public IP and which port has been opened by NAT to permit incoming traffic to enter the client's network. This information enables the Phoenix ALIO to identify its position within the SIP server. This protocol is used in "AUTO3" and "AUTO4" NAT TRAVERSAL modes (see section 4.3).



Registration in a STUN server



Notification from the server of the corresponding NATs detected

The response further enables the STUN client to determine the type of NAT being used, since different NAT types handle incoming UDP packets in different ways. STUN supports three of the four main existing types of NAT: *Full Cone*, *Restricted Cone* and *Port Restricted Cone*. It does not, however, support *Symmetric* NAT, also known as bidirectional NAT, although TALENT allows it to be detected and reports its presence to the user.

Once the client has discovered its public address, it can advise its peers of that address.

STUN is useful as a complement to protocols like SIP. SIP uses UDP packets to signal sound, video and text traffic over the Internet, but does not enable communication to be established when the devices at the ends of the communication circuit are behind NAT routings. This is why STUN is customarily used in these applications, to permit communication to be established.

The connection with the STUN server is normally made through port 3478 by means of UDP. The STUN server can then provide the client with an alternate IP and communication port.

For complete information on the STUN protocol, we recommend consulting:
<http://tools.ietf.org/html/rfc3489>

AEQ always has a PHOENIX unit available for test at "phoenixMaster@sip.aeq.es" URI and its 2 SIP servers are also available at sip.aeq.es and sip2.aeq.es and with warranty that both work according to the official standard.

APPENDIX C: Ports used by TALENT.

When a TALENT is installed in a private IP network and you want to establish communication with other units through that network router+firewall (gateway), three indications related to the ports used by the unit must be taken into account:

1 - Output permissions in router+firewall.

TALENT will send packets to different servers and/or other units (each one will use a different port). Therefore, firewall will have to allow that packets from TALENT or Phoenix IP address are sent towards the following ports:

Protocol	Port number	Usage
UDP	53	DNS (domain name server) protocol
UDP	162	SNMP Traps port
UDP	5004-5010 (may change)	RTP protocol (audio packets going towards the remote unit, ports depend on the remote unit network, not Phoenix one)
UDP	3478-3479	STUN protocol (server for getting the public IP of the private network)
UDP	5060-5062	SIP protocol (SIP signalling packets, port depends on the remote unit network, not Phoenix network)
UDP	50000-60000	Vumeters protocol (only when remote control PC is not in Phoenix network)
TCP	4422	Remote control protocol (only when Phoenix must be connected to PC and PC is not in its network. Port depends on PC, not on Phoenix)

2 - Input permissions in router+firewall.

TALENT will have to be able to receive packets from units installed out of the private network. Therefore, firewall will have to allow that packets sent to TALENT or Phoenix IP address to the following ports are received:

Protocol	Port number	Usage
UDP	161	SNMP protocol (monitoring)
UDP	5004-5010 (may change)	RTP protocol (audio packets towards the remote unit, ports depend on Phoenix configuration)
UDP	5060-5062	SIP protocol (SIP signalling packets, port depends on the remote unit network, not Phoenix network)
TCP	80	HTTP protocol (only when PC must be connected to Phoenix web server for a firmware upgrading, for instance)
TCP	4422	Remote control protocol (only when PC must be connected to Phoenix and PC is not in its network)

3 - When router+firewall uses NAT translation between private and public addresses, then a Port Forwarding must be made in the router for each one of the ports described in section 2 and for each one of the IPs of TALENT or Phoenix units installed in that private network. In that case, remote unit will send its packets towards router IP (it's a public IP) and the ports configured in router by means of Port Forwarding (these are public ports), not to each TALENT or Phoenix ports (these are private ports).

APPENDIX D: Application notes guide.

This index tries to give users guidance on selecting the most advisable application note in order to connect two audiocoders of Phoenix family, depending on its requirements and working environment. Each application note describes the way to configure each of the audiocoders. When both ends are different (for instance, at one end there's a Phoenix Mobile and at the other end a Phoenix ALIO), different application notes should be followed in order to configure each one. All notes are available in electronic format on the AEQ website.

D1. Internet connection using standard cable access.

Application note AN0A. Phoenix Studio, Venus or Mercury audiocoder directly connected to Internet by means of a dedicated cable-Modem with DHCP. SIP call using AEQ SIP Proxy.

Application note AN0B. Same case as AN0A, but adapted to Phoenix Mobile.

Application note AN0C. Phoenix Studio, Venus or Mercury audiocoder connected to a LAN (together with other IP devices), connected to Internet by means of a router with NAT that can be configured (or we have access to the Network Manager). SIP call using AEQ SIP Proxy.

Application note AN0D. Same case as AN0C, but adapted to Phoenix Mobile.

Application note AN0E. Same case as AN0C, but calling in DIRECT SIP with no SIP Proxy involved.

Application note AN0F. Same case as AN0E, but adapted to Phoenix Mobile.

Application note AN0G. IP Audio distribution at large international events using Phoenix Studio audiocoder.

Application note AN0H. AoIP distribution and contribution system with Phoenix Venus audiocoders for radio broadcasting companies.

Application note AN0I. Login in the alternative AEQ server if working with SIP connectivity some trouble situations are faced.

D2. Special applications using different kinds of Internet physical accesses, or point to point connections.

Application note AN1. Connecting a Phoenix (Studio, Venus, Mercury or Mobile) to Internet through a PC via a WiFi network.

Application note AN2. Connecting two Phoenix Mobile units using a BGAN satellite link.

Application note AN3. Connecting two audiocoders (Phoenix Studio - Phoenix Studio & Phoenix Mobile - Phoenix Studio) using a private WiMAX network.

Application note AN4. Connecting two Phoenix Studio units using a dedicated point-to-point IP radio link.

Application note AN4B. Same case as AN4, but also for Venus and Mercury and configuring the units via remote control software.

Application note AN5. Connecting Phoenix IP (or TALENT) audiocoders to 3G/4G networks.

Application note AN6. Connection and remote control of Phoenix Alio (or Talent) audiocoders from the studio and simultaneous usage as a phone set.

Application note AN7. 3G/4G wireless transmission system using AEQ Talent or Alio audiocoders.

APPENDIX E: TALENT Troubleshooting guide.

E1. TALENT unit won't turn on or the blue power indicator blinks.

If there is no light at the unit's top blue indicator, please make sure that the connected power supply is properly selected using the rear "POWER SWITCH" (DC-IN → switch in / USB power → switch out) and that the selected power source actually provides proper voltage (5V at USB or 5-15V at DC IN).



In case that you are supplying power to the TALENT using a USB cable, a blinking blue power LED means that the power source in use is not able to provide enough current (this may happen with certain laptops or small phone chargers), so you should select another power supply method.

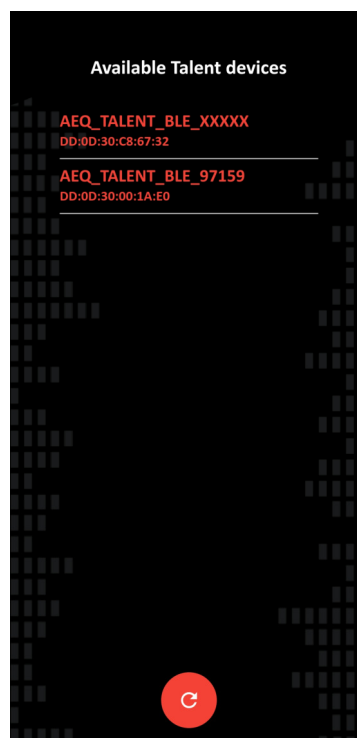
We always recommend the usage of the supplied AC/DC adapter, when possible.

E2. I cannot control TALENT from "AEQ TALENT PILOT" App.

Make sure that TALENT is not yet controlled by another smart device. Turn Bluetooth on in your phone and check that, after a few seconds, the following devices are found:

- "AEQ_TALENT_<serial number>"
- "AEQ_TALENT_BLE_<serial number>" ← this second device is the one used for control App.

Once you have checked that your phone has found both devices, close "AEQ TALENT PILOT" App and open it again after a few seconds, or touch the bottom Refresh red button. Your TALENT should then appear in the list:



so after touching on its name, you should be able to control it (in order to verify that, once in the main screen touch the ON AIR button, for example, and check that changes are made in parallel in the screen and on the actual unit).



If you find problems with "AEQ TALENT PILOT" App, make sure that your TALENT unit is updated to CPU version **9.30** or above. You can check this in "AEQ ControlPHOENIX" under Config → Miscellaneous → Firmware versions. If your firmware is older, please contact AEQ Technical Service to get a free upgrade.

E3. I can control TALENT but I am not able to send it Bluetooth audio.

As described in E2, check that both devices have been found and try to pair the one named "AEQ_TALENT_<serial number>" (without "_BLE"). After providing the required confirmations, the phone should say "Connected for Calls and Multimedia Audio" or something similar. The TALENT's blue Bluetooth LED should get steadily illuminated (it will actually look pinkish if "AEQ TALENT PILOT" App is already connected).



If the unit is properly linked but you cannot hear any audio, please first make sure that your phone output level is up. You can do this from the phone itself or using "AEQ TALENT PILOT" App: if you touch the left audio button, the slider appearing at the right corresponds to that output volume. Check that this same input is ON AIR if you want to send it to Program (in that case, audio should be displayed in the TALENT's and App vu-meter) and also check that the corresponding headphone or auxiliary outputs (accessible by touching the right audio button in the main screen) have their LINE IN source selector active for local pre-listen.



E4. LAN Indicator is blinking.

If the yellow network symbol is blinking, then not all requirements to initiate a call in the currently selected mode (RTP, DSIP or PSIP) are met.



Please verify that the Ethernet cable is properly connected, at first place. The green LED at that connector itself should blink. If the unit is configured in DHCP (that's how it is from factory) then maybe your router or network doesn't have an active DHCP server and hence TALENT is unable to obtain valid IP configuration. Check current IP configuration with "AEQ TALENT PILOT", for example.



If all the above is correct and you are in PSIP mode but your yellow network indicator is still blinking, then the unit is unable to register in the selected SIP server. Make sure that Internet is reachable from that port (for example, by checking on another device connected to the same network) and, if it is, check your SIP account settings (user, password and provider) using "AEQ ControlPHOENIX" or "AEQ TALENT PILOT".

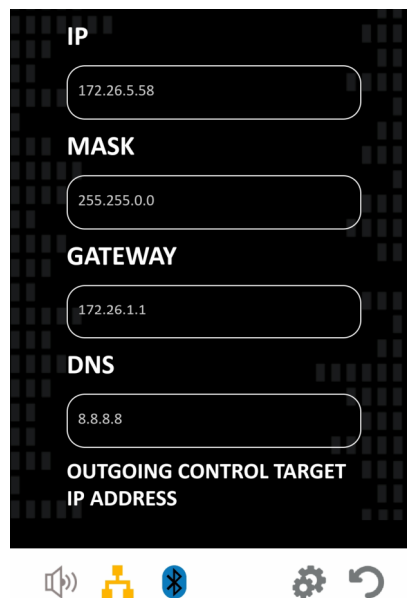


You can always verify TALENT's current configuration using "AEQ TALENT PILOT" App, entering the Adjustments menu (gear icon). There you can check whether your unit is in PSIP mode, touch Settings to check IP Settings (verify that DHCP mode is active if you haven't changed to fixed IP addressing and that valid IP configuration has been got). And you can also go back and touch SIP Settings to verify your SIP account settings.

E5. I am not able to control TALENT from "AEQ ControlPHOENIX" in my local PC.

If you are not able to find TALENT during "AEQ ControlPHOENIX" start-up auto discovery, make sure that the PC and TALENT codec are configured in the same network.

You can check TALENT's correct IP configuration using "AEQ TALENT PILOT" App by entering the Adjustments menu (gear icon), touching Settings → IP Settings. If it is configured with DHCP ON, it should have valid IP data, unless DHCP is not working. Then you can try entering a valid IP Address, Mask, Gateway and DNS manually.



If you need to go back to a known IP address, even without the App, press the rear "RESET" button (don't do this if you are in the middle of a call).. Press and release it only when you hear two beeps. The unit will reboot and adopt the default IP configuration: 192.168.1.89 / 255.255.255.0.

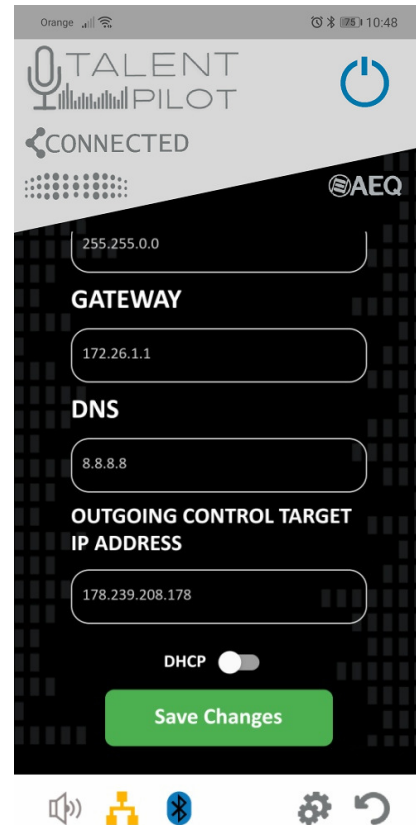
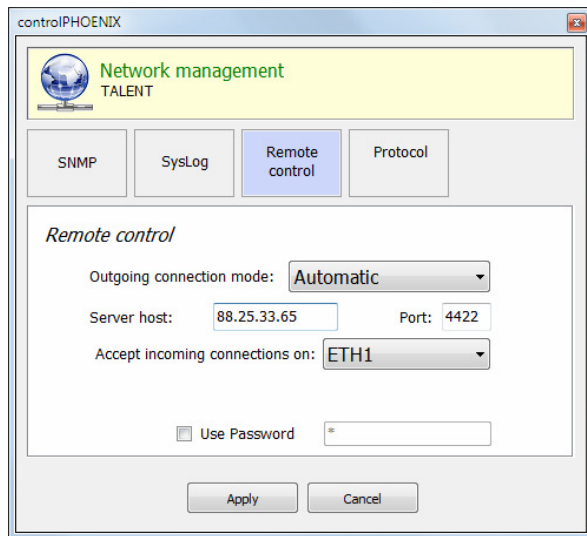
You can always check for connectivity by opening a CMD window in your PC and executing "ping 192.168.1.89" or typing this address in your browser's URL bar.

Try to manually add the TALENT to "AEQ ControlPHOENIX" using this IP (Equipment → New).

E6. I want a remote operator to control my TALENT remotely using "AEQ ControlPHOENIX" (Advanced Topic).

If you have Internet connection, you can have your TALENT controlled from the remote studio operators, provided that they have "AEQ ControlPHOENIX" installed. In order to do that, check with them whether this software is available (with version above 2.2.1.47) and ask them for their public IP address. This will require previous configuration in the studio's router (which they will have probably performed already if they have AEQ audiocodecs). At your side, you only need to configure TALENT so it points to the proper studio's public IP address for outgoing control. In order to do that, you have two alternatives:

- 1) Using "AEQ ControlPHOENIX" in your local PC, go to *Configuration* → *Network* → *Remote control*, and establish the "Outgoing connection mode" as Automatic. Next field, "Server host", should be filled in with the public IP address indicated by the studio.
- 2) Using "AEQ TALENT PILOT" App, go to *Adjustments* (gear icon) → *Settings* → *IP Settings*. If you scroll to the bottom, there is an option to establish the outgoing IP address provided by the studio.



E7. I am able to make a call but the "CONNECTED" indicator is blinking.

If that's your case, your TALENT is not receiving any audio data back from the remote end.

- If the unit is configured in RTP mode, this may mean that the remote codec isn't receiving your data either, or that it is receiving them but it's not sending return properly or it's simply not reaching TALENT. Verify that the destination **IP:port** set when making the call are reachable (ask the remote peer whether its IP is public and the port is properly forwarded to the remote codec, which should be correctly configured as well, especially if it is not from AEQ)
- If you TALENT is configured in SIP mode, then the call has been coursed and answered but audio data is not received back. Verify NAT Traversal configuration using "AEQ ControlPHOENIX" or contact your IT support or AEQ Technical Service for a solution.

E8. I am not able to make TALENT register in AEQ default SIP server.

There are several reasons why a codec may not be able to register into a server. If you are using the default account provided by AEQ for your unit from factory, you only need to have Internet connectivity and set the device to PSIP mode. In this case, the yellow network indicator should illuminate steadily.

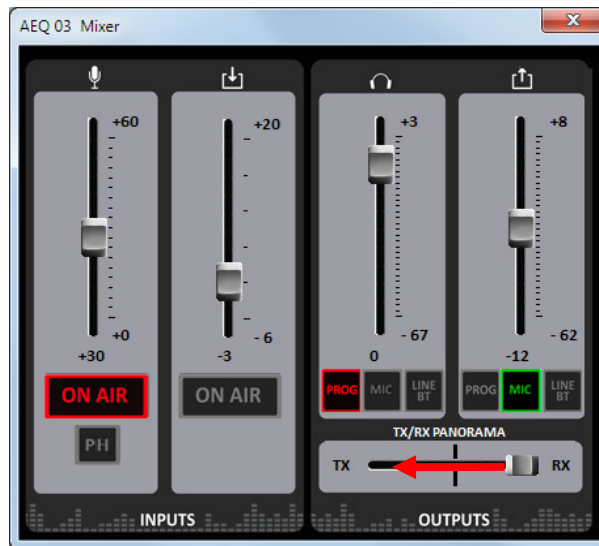


If it is not registering, please check IP configuration, and if you are using DHCP, check that you have obtained a valid IP configuration (see paragraph E5). If, on the other hand, you are not using the SIP account provided by default, please check the server address and account data carefully using "AEQ ControlPHOENIX" or "AEQ TALENT PILOT".

However, even when correct configuration is made, we have recently observed that some ISP forbid the use of standard SIP ports in their routers (usually in the 5060-5062 range). Check with your IT support if you face this situation.

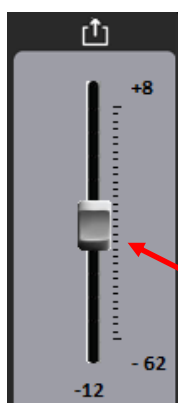
E9. A call has been established with the studio. They can hear me but I can't hear myself.

Many studios don't return audio in order to avoid talker's confusion due to the back and forth delay. In this case, you can configure TALENT to provide you with "local feedback" directly from your microphone to your headphones, so you can hear yourself without delay. In order to do this, activate the "MIC" source in your headphone output using "AEQ ControlPHOENIX" mixer or "AEQ TALENT PILOT" output audio adjustments. Alternatively, if PROG is selected as one of the sources for your headphones (this will usually be the case, as you want to hear studio return), you can adjust the "TX/RX PANORAMA" slider to provide some program-go feedback (TX) to your headphones. You can also do this from TALENT's front panel by pressing the left encoder button and then turning the knob to the left until you get the desired feedback level.



E10. I want to make a phone call to an interviewee and want to send it to the studio.

In order to do this, first of all pair **Bluetooth** audio. See section 6.5.



Next, make sure that your MIC is sent to LINE OUT/BT (you can simply long-press the right encoder and its associated red light will become illuminated).



Adjust Bluetooth send level by briefly pressing the same encoder (its associated yellow light will become illuminated) and adjust the sending level with the knob.



We recommend a moderate value of Line OUT/BT output level, below 0dB to avoid saturating the terminal so your remote peer can hear you clearly.

On the other hand, you have to make sure that LINE IN is routed to the headphones (if you want to hear the remote peer), or, additionally, that LINE IN is ON AIR when you are ready to send the interviewee voice to the program.

Remember that you can control whether Bluetooth incoming audio is sent to program by means of a long pressing of the left encoder (the associated red light will become illuminated to indicate that this input is ON AIR).



For advanced applications like this, it is always useful to have "AEQ ControlPHOENIX" mixer or audio adjustments in the App open to check the send/receive listening levels properly.

APPENDIX F: Additional information.

NOTE: This equipment complies with the limits for a Class A digital device, pursuant to part 15 of the **FCC** Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.