



AEQ PHOENIX MOBILE



USER'S MANUAL

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**Firmware Versions: Micro 3.22 / DSP 3.39 / FPGA 1.47 or higher
PGA-01 POTS Module: Micro 1.05
PGA-03 ISDN Module: Micro 1.01 / TA 6.00**



A.E.Q., S.A., the company that manufactures this equipment, is an AENOR “registered company” under number ER-080/1/96, in compliance with ISO - EN - 9001 standard.

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1. INTRODUCTION.

1.1. General overview.

The PHOENIX MOBILE is an IP audio codec for mobile applications. It includes a digital mixer with four analog inputs, an optional LI-ION battery, a 3.5" multifunction display screen and a 12V DC battery charging power unit.

PHOENIX MOBILE is a portable codec over IP fully compatible with the EBU-TECH 3326 technical specification formulated by the EBU N/ACIP working group, which was developed to guarantee compatibility between different manufacturers in applications for contributing professional quality audio over IP networks.

In addition to the Ethernet connection, the device has two extra slots for inserting additional communication modules (POTS and ISDN type in this version) and an USB port that allows you to connect a mobile phone in order to establish a GSM or 3G communication. This modular design makes the PHOENIX MOBILE easy to update in the future with new communication modules responding to the needs the market may demand at any time.

Besides the program signal and its corresponding return, the PHOENIX MOBILE can simultaneously provide a channel for redundancy or coordination purposes using the additional communication modules.

The PHOENIX MOBILE is designed to ensure interoperability with existing and future communications equipment developed by AEQ and other manufacturers, thanks to the large number of coding algorithms it implements, and to the possibility of easily update as the market evolves.

AEQ recommends the complete reading of the present manual before the utilization of the unit, though in the chapter 4 an user's quickstart guide is available.

1.2. Functional specifications.

- Full-duplex channel management (Program and Coordination) over as many as four possible communication interfaces: Ethernet (native), USB and two optional communication modules.
- Inputs:
 - Two microphone inputs (via XLR3-Female) with preamplifier and phantom feed option.
 - A selectable MIC/LIN input (via XLR3-Female) with preamplifier.
 - An Auxiliary line input (via XLR5-Female).
- Outputs:
 - Two stereo headset outputs (via ¼" Jack).
 - An Auxiliary output (via XLR5-Female).
- Complete, integrated router and digital mixer.
- IP communication interface (Ethernet).
- PHOENIX MOBILE implements SIPv2 according to RFC-3261 and RFC-3264.
- IP and GSM communication interface (USB).
- Optional POTS communication interface (PGA-01), equipped with a telephone modem that offers a connection like a telephone hybrid or like a data communications device for high quality transmission of audio signals over POTS lines. Includes a frequency extender option.
- Optional ISDN communication module (PGA-03) for 2 B channels (2 x 64Kbps).

To simplify the operation of the unit over IP, AEQ offers without extra cost the services of a proprietary SIP server. For more information, consult annexe C4 of the present manual.

- The AEQ Phoenix Mobile includes a protective plastic cover which allows access to the audio mixing controls while avoiding erroneous operation due to inattention of the rest of the critical communication functions.
- Absolute control of all the internal routing of audio signals, allowing from basic mono or stereo routing configurations to complex configurations for outdoor applications involving reporters, commentators, producers, technicians, etc.
- This equipment can receive and decode stereo audio, generating a monophonic signal for internal distribution.
- Includes connections for an universal 12V or 15V DC battery charger and power supply.
- Optional rechargeable and easily extractable LI-ION battery. Also incorporates handle function for carrying.

1.3. Available encoding modes.

The encoding modes available vary according to the communication interface or module selected to establish the connection.

We recommend your consulting annexe A to see the detailed description and additional information on each of the modes.

AVAILABLE ENCODING MODES IN EACH COMMUNICATION INTERFACE

Program Channel

Encoding modes	IP (Ethernet and USB-3G)	POTS	ISDN	GSM (USB-GSM)
ITU G.711 A-Law mono	√		√	
ITU G.711 μ-Law mono	√		√	
G.711 A-Law mono EXTENDED	√		√	
G.711 μ-Law mono EXTENDED	√		√	
ITU G.722 Statistical Mono	√		√	
AEQ LD EXTENDED 128 Kbps	√		√	
MPEG-2 LII 128 mono 24 KHz	√		√	
MPEG-1 LII 128 mono 32 KHz	√		√	
MPEG-1 LII 128 mono 48 KHz	√		√	
MPEG-1 LII 128 estéreo 32 KHz	√		√	
MPEG-1 LII 128 estéreo 48 KHz	√		√	
MPEG-2 LII 64 mono 16 KHz	√		√	
MPEG-2 LII 64 mono 24 KHz	√		√	
MPEG-1 LII 64 mono 32 KHz	√		√	
MPEG-1 LII 64 mono 48 KHz	√		√	
PHONE		√		
PHONE (AEQ frequency extender)		√		
MPEG-4 AAC-LC 12 Kbps mono 24 KHz		√		
MPEG-4 AAC-LC 22 Kbps mono 24 KHz		√		
MPEG-4 AAC-LC 32 Kbps mono 24 KHz	√			
MPEG-4 AAC-LC 32 Kbps mono 48 KHz	√			
MPEG-4 AAC-LC 64 Kbps mono 24 KHz	√		√	
MPEG-4 AAC-LC 64 Kbps mono 48 KHz	√		√	
MPEG-4 HE-AAC 9 Kbps mono 48 KHz				√
MPEG-4 HE-AAC 12 Kbps mono 48 KHz		√		
MPEG-4 HE-AAC 22 Kbps mono 48 KHz	√	√		
MPEG-4 HE-AAC 32 Kbps mono 48 KHz	√			

Coordination Channel

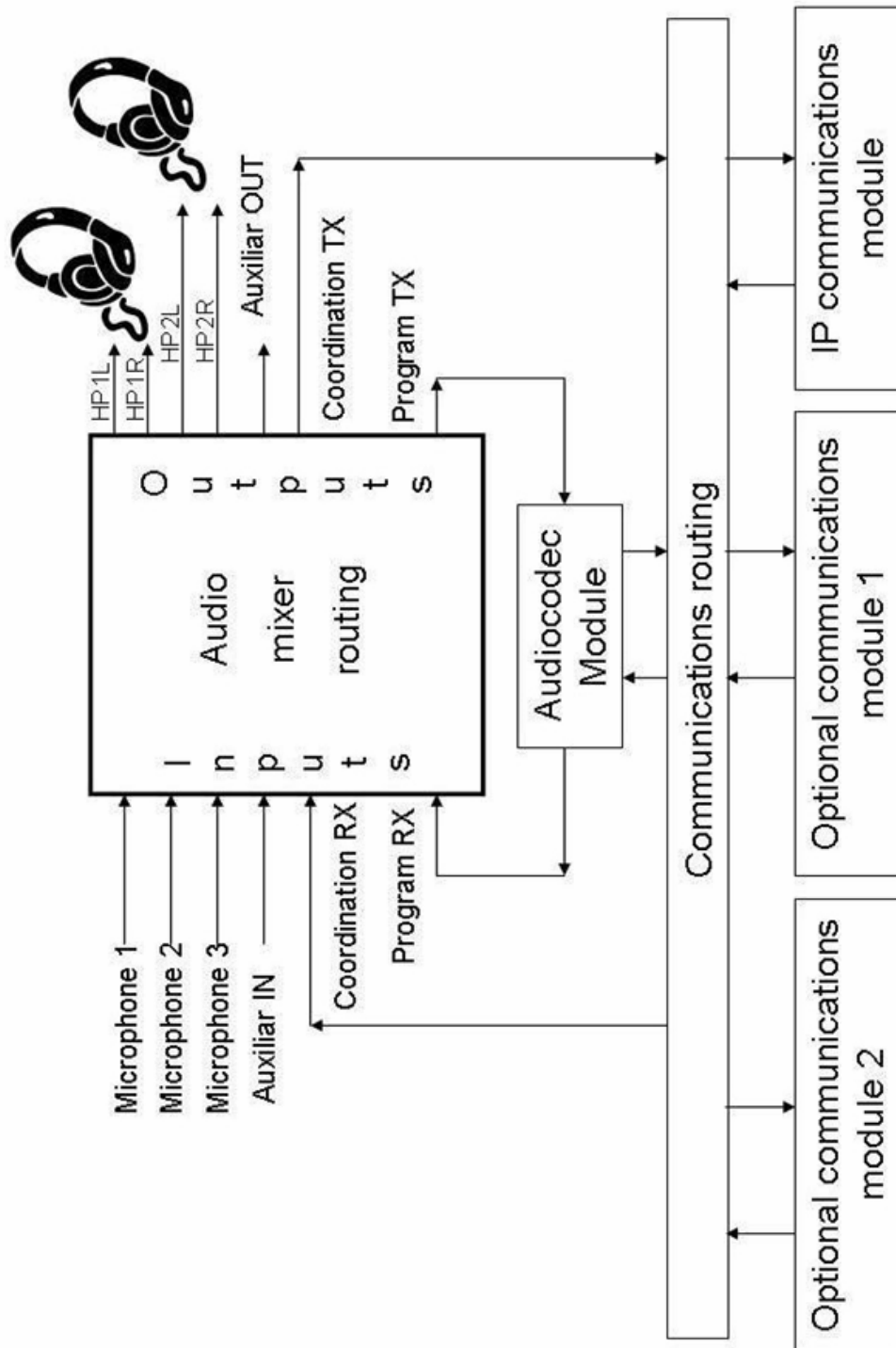
Encoding modes	IP (Ethernet and USB-3G)	POTS	ISDN
G.711 A-Law mono	√		√
G.711 μ-Law mono	√		√
G.711 A-Law mono EXTENDED			√
G.711 μ-Law mono EXTENDED			√
PHONE		√	

NOTE: It's not possible to establish a **GSM** communication when the mobile phone connected to USB port is associated to Coordination channel.

CHARACTERISTICS OF THE ENCODING MODES

Encoding modes	Bit rate (Kbps)	Bandwidth (KHz)	Delay
PHONE	Uncoded	3.4	Very low
PHONE (AEQ frequency extender)	Uncoded	3.1	Very low
ITU G.711 A-Law mono	64	3.5	Low
ITU G.711 μ-Law mono	64	3.5	Low
G.711 A-Law mono EXTENDED	64	3.5	Low
G.711 μ-Law mono EXTENDED	64	3.5	Low
ITU G.722 Statistical Mono	64	7	Low
AEQ LD EXTENDED 128 Kbps	128	15	Low
MPEG-2 LII 64 mono 16 KHz	64	7.5	Medium
MPEG-2 LII 64 mono 24 KHz	64	11.25	Medium
MPEG-1 LII 64 mono 32 KHz	64	10.5	Medium
MPEG-1 LII 64 mono 48 KHz	64	10.5	Medium
MPEG-2 LII 128 mono 24 KHz	128	11.25	Medium
MPEG-1 LII 128 mono 32 KHz	128	15	Medium
MPEG-1 LII 128 mono 48 KHz	128	20	Medium
MPEG-1 LII 128 estéreo 32 KHz	128	10.5	Medium
MPEG-1 LII 128 estéreo 48 KHz	128	10.5	Medium
MPEG-4 AAC-LC 12 Kbps mono 24KHz	12	3.375	High
MPEG-4 AAC-LC 22 Kbps mono 24KHz	22	5.625	High
MPEG-4 AAC-LC 32 Kbps mono 24KHz	32	6.750	High
MPEG-4 AAC-LC 32 Kbps mono 48KHz	32	16	High
MPEG-4 AAC-LC 64 Kbps mono 24KHz	64	11.520	High
MPEG-4 AAC-LC 64 Kbps mono 48KHz	64	20	High
MPEG-4 HE-AAC 12 Kbps mono 48KHz	12	10.875	High
MPEG-4 HE-AAC 22 Kbps mono 48KHz	22	15.375	High
MPEG-4 HE-AAC 32 Kbps mono 48KHz	32	16.875	High

1.4. Block diagram.



For further information, see complete diagrams in annexe B.

1.5. Compatibility with other AEQ codecs.

Through the use of the additional communication modules, the PHOENIX MOBILE offers its users the option of connecting with other AEQ equipment of previous generations.

The POTS module enables its users to communicate with other devices that include telephone hybrids such as TH02-EX, TLE02, TLE02D MPAC, SWING and COURSE hybrid boards, in addition to offering the added possibility of a frequency extender.

The ISDN module enables its users to communicate with other devices that include ISDN interface such as TLE02D, SWING, MPAC, EAGLE and COURSE ISDN boards.

1.6. Compatibility with other manufacturers' codecs.

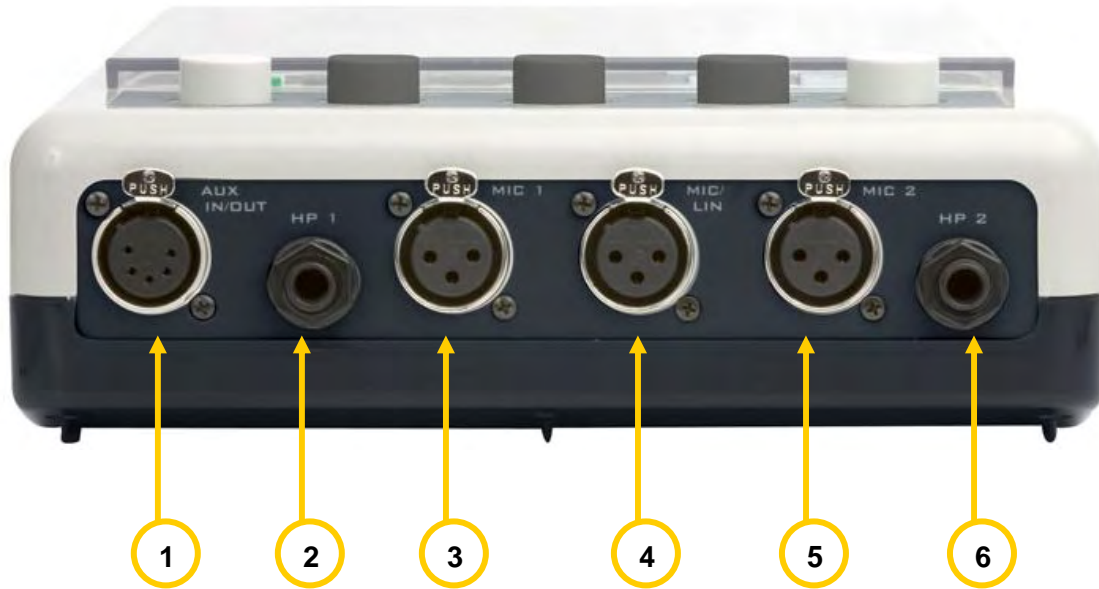
PHOENIX MOBILE is a portable encoder over IP, fully compatible with the EBU-TECH 3326 technical specification formulated by the EBU N/ACIP working group, which was developed to guarantee compatibility between different manufacturers in applications for contributing professional quality audio over IP networks. Therefore, the PHOENIX MOBILE can be connected with any codec made by another manufacturer with guaranteed connection over IP, provided that the other unit was developed according to N/ACIP (see technical specifications).

With the rest of communication modules and interfaces, compatibility will depend essentially on the coding algorithms implemented by the manufacturers of the other equipment, since the signaling part is practically universal. This is why we recommend consulting, before operating with other manufacturers' equipments, the technical characteristics of those equipments.

2. PHYSICAL DESCRIPTION OF THE UNIT.

To understand how the unit is wired and installed, you will first need to familiarize yourself with the connectors and other configurable elements that are present on the front and rear panels of the device.

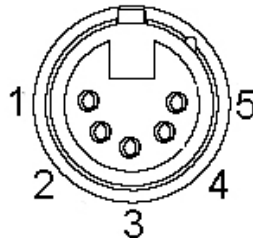
2.1. Description of the front panel and connections.



2.1.1. Auxiliary I/O ("AUX IN/OUT").

1

XLR - 5p - Female connector. Balanced connection.



XLR 5p Female-panel pinout

Pin 1 → Common ground

Pin 2 → + Input

Pin 3 → - Input

Pin 4 → + Output

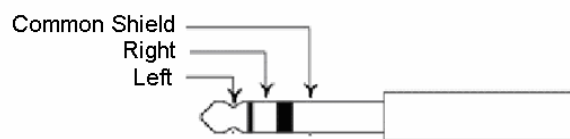
Pin 5 → - Output

2.1.2. Headphones 1 and 2 output ("HP 1" and "HP 2").

2

6

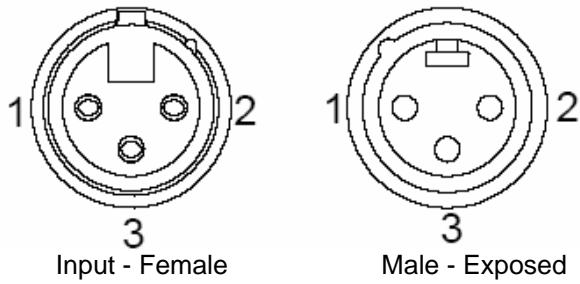
¼" TRS Jack connector. Asymmetrical connection.



TRS Jack pinout

2.1.3. Microphones 1 and 2 input (“MIC 1“ and “MIC 2“). 3 5

XLR - 3p - Female connector. Balanced connection.



Connector viewed from the soldered part.

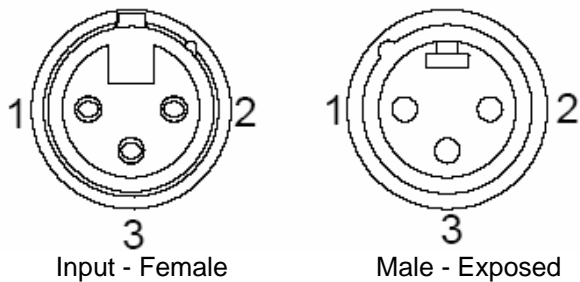
XLR 3P pinout

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

2.1.4. Mic-Line input (“MIC/LIN“). 4

Selectable via software.

XLR - 3p - Female connector. Balanced connection.



Connector viewed from the soldered part.

XLR 3P pinout

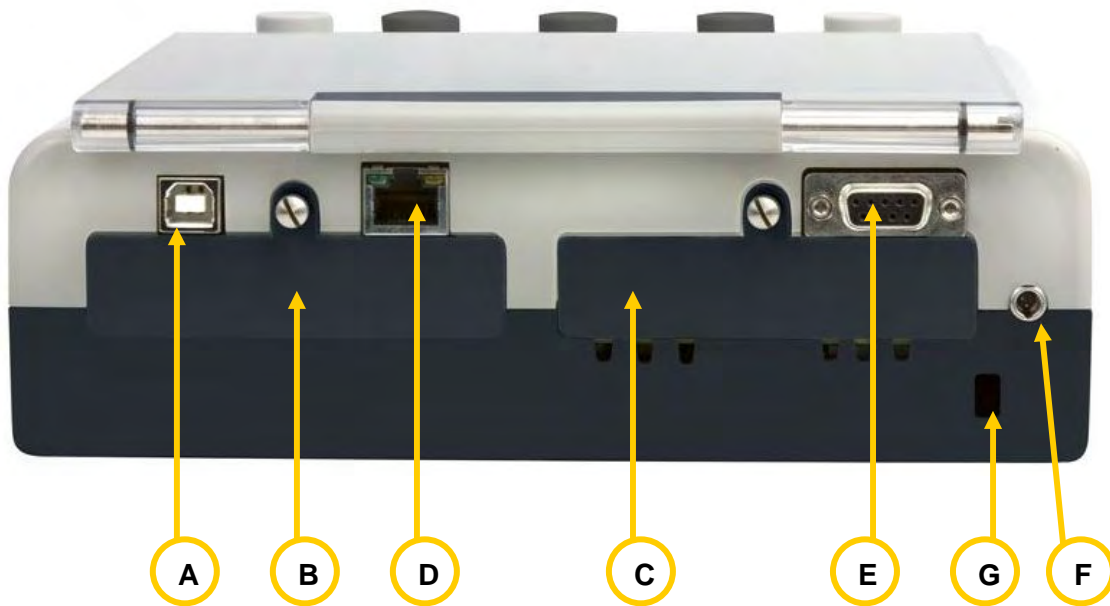
Pin 1 → Ground
 Pin 2 → + Input
 Pin 3 → - Input

Detailed explanation:

The microphone inputs (MIC1, MIC2) include low-noise preamplifiers and supply Phantom power feed (+12VDC and 10mA), which can be activated via software, to enable both dynamic and electrostatic microphones to be connected.

The level of input signals MIC1, MIC2 y MIC/LINE and the level of output signals HP1 and HP2 are modifiable through the rotary encoders associated with each signal on the control surface.

2.2. Description of the rear panel and connections.



2.2.1. USB Port. **A**

The USB port allows you to connect a mobile phone (by means of mobile phone data cable and an USB B to UBS A adapter) in order to establish a **GSM** or **3G** communication.

The USB port can be configured as Master or Slave. By default, it leaves the factory configured as Slave. This port can be connected to a USB port in a PC, permitting a data transmission speed of up to 12 Mbps (USB 1.1).

2.2.2. Right additional communication module. **B**

Right slot when the codec is viewed from the front. This right slot name is the identifier used in the configuration menus to refer to this additional communications module.

While PHOENIX MOBILE is supplied by default with that slot empty and a blank cover, any additional communications module can be inserted there as an option: PGA-01, PGA-02, PGA-03 or any other module AEQ may develop in the future. At the time this manual was written, PGA-01 POTS and PGA-03 ISDN communication modules are available.

The installation of any additional module presents no difficulty: turn the device off, remove the blank cover, insert the new module and turn the equipment on. The unit will automatically detect the presence of the new communications interface.

2.2.3. Left additional communication module. **C**

Left slot, with the codec viewed from the front. This left slot name is the identifier used in the configuration menus to refer to this additional communications module.

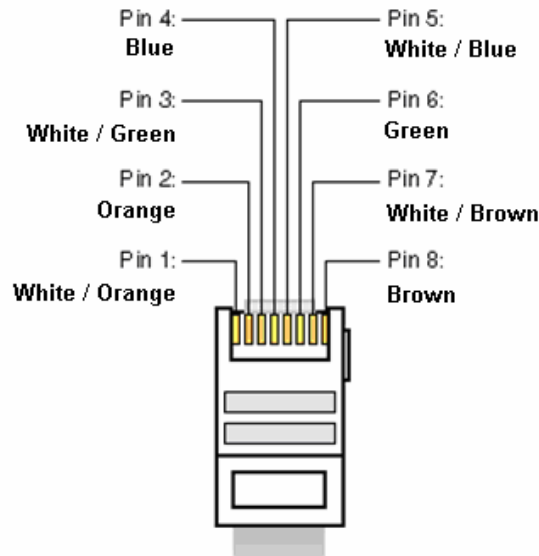
While PHOENIX MOBILE is supplied by default with that slot empty and a blank cover, any additional communications module can be inserted there as an option: PGA-01, PGA-02, PGA-03 or any other module AEQ may develop in the future. At the time this manual was written, PGA-01 POTS and PGA-03 ISDN communication modules are available.

The installation of any additional module presents no difficulty: turn the device off, remove the blank cover, insert the new module and turn the equipment on. The unit will automatically detect the presence of the new communications interface.

2.2.4. Ethernet Port. **D**

Using this Ethernet port will enable you to connect the unit via IP to a LAN or WAN network and send audio over it. This port can also be used to receive additional data from a PC.

Physically it is an RJ45 10 BT connector.



RJ45 connector pinout

2.2.5. GPIO connections. **E**

This port will enable a basic external control of the unit. This function is not implemented in this firmware version.

NOTES:

- The output signals are open collector, and are active at low level.
- The input signals are high level and only active in the presence of pulse edges.

The connector used is a DB9 female, with the following pinout:

- pin 1: Ground
- pin 2: GPI1 -
- pin 6: GPI1 +

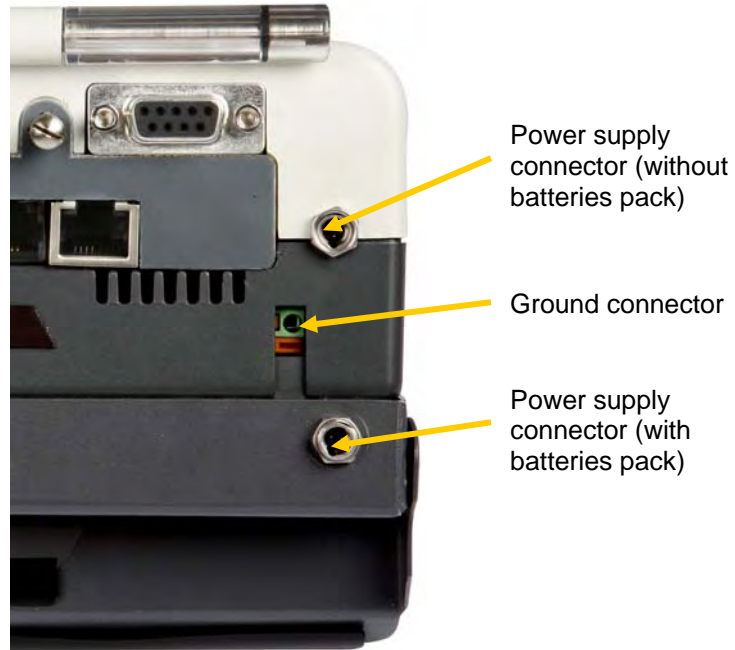
2.2.6. Power Supply.

2.2.6.1. General.

The equipment was designed for use while hanging from a shoulder strap or resting on a table top. As an option, it can include a removable Li-Ion batteries pack (3500 or 4000mAh / 7.4 V), which will power the codec for an average of more than 2.5 hours in normal operating mode, integrating carrying handle.

F The power supply connector (mini-XLR3p female) is located on the back panel of the codec. **IMPORTANT NOTE:** In case of a batteries pack is connected to the unit, this connector is no longer operative and the one that's included in the batteries pack itself must be used.

G The equipment also has a ground connector.



Power supply connection details

The power switch (push button) is located on the left side of the unit.

2.2.6.2. Power supply / Battery charger.

The AEQ PHOENIX MOBILE can be supplied with electric power from an external battery charger / power feed unit specifically designed for this purpose through the 12V DC-INPUT connector.

The equipment must be powered with the battery charger supplied with the unit.

The charger cable termination is a mini-XLR 3p female connector with a secure anchoring device to prevent accidental disconnections.



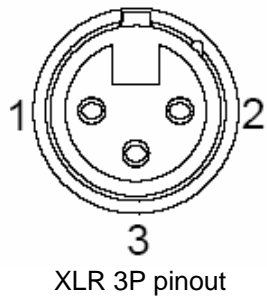
Mini XLR 3p Female connector

The power supply / battery charger supplied is of the universal type, and accepts voltages ranging from 100 to 240 VAC / 50-60 Hz.



Universal charger and power feed cable supplied

Pinout of the mini-XLR 3p female battery charger connector:

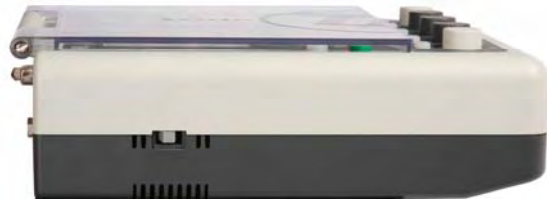


Connector viewed from the soldered part

Pin 1 → Ground
Pin 2 → V +
Pin 3 → Not used



PHOENIX MOBILE with battery



PHOENIX MOBILE without battery

2.3. Description of the control surface.

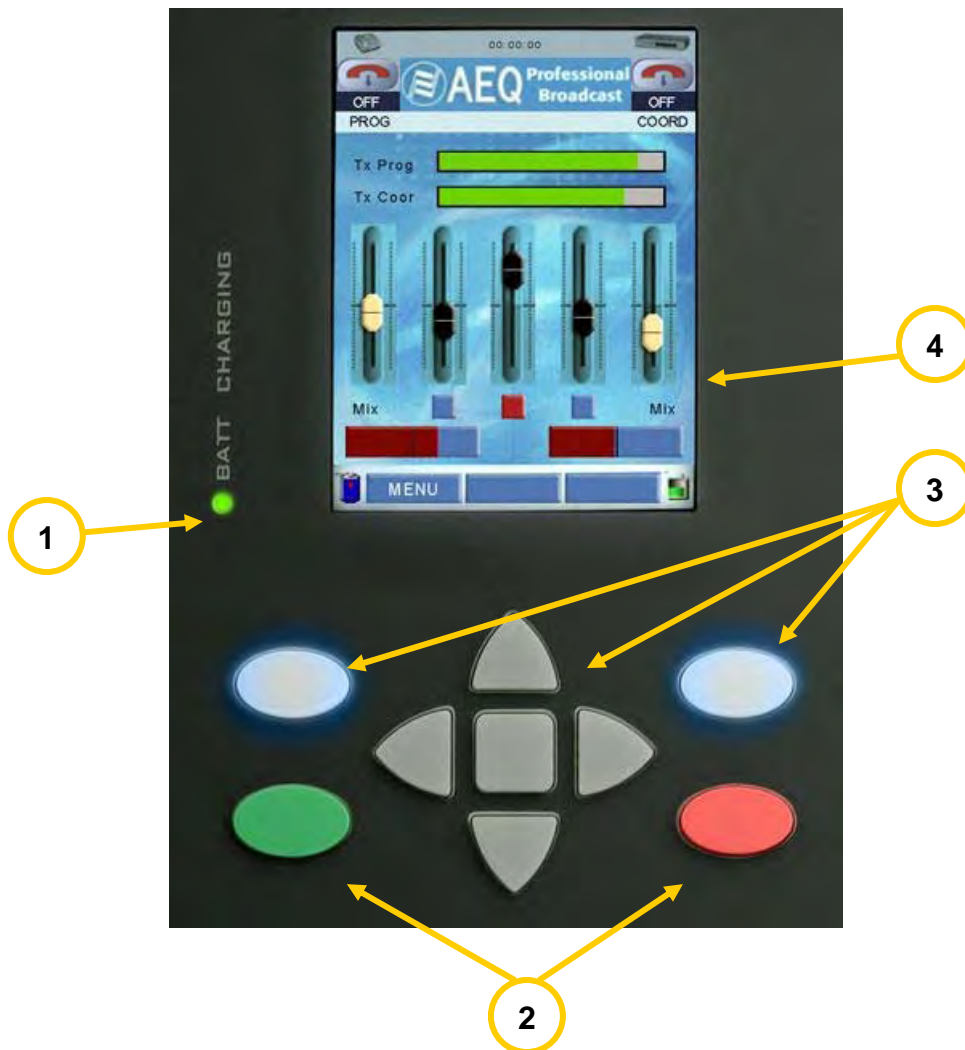
The following paragraphs describe the various functional areas that make up the control surface.



1. Multifunction display screen and navigation keys.
2. Alphanumeric keypad.
3. Function and routing keys and communication status LEDs.
4. Input channel mixing controls, headphone volume control and channel activation keys

2.3.1. Multifunction display screen and navigation keys.

3.5" TFT screen with five navigation keys to allow the user to move through the operation and configuration menus. Also included are two keys to accept / cancel the current operations in the menus, and two keys to call/offhook/hangup telephone calls made or received by the user.



2.3.1.1. Battery charging indicator. 1

For further information, please consult section 4.2 in this manual.

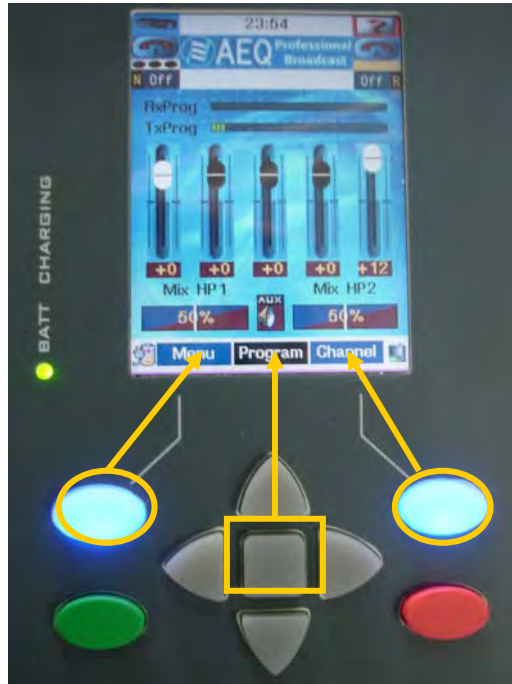
2.3.1.2. On-hook / Off-hook keys. 2

Allow you to take the currently active call off-hook (green button) and place it on-hook (red button) instantaneously.

2.3.1.3. Navigation keys.

3

The four arrows let you navigate in the on-screen menus: scrolling “↑” and “↓” in a menu, and laterally “←” and “→” to select a value. The left and right keys (lighted in the figure) and the center key enable you to select among the various options that will be displayed in the lower part of the screen; their functions will change according to the context.



Association of the navigation keys with icons displayed on multifunction screen.

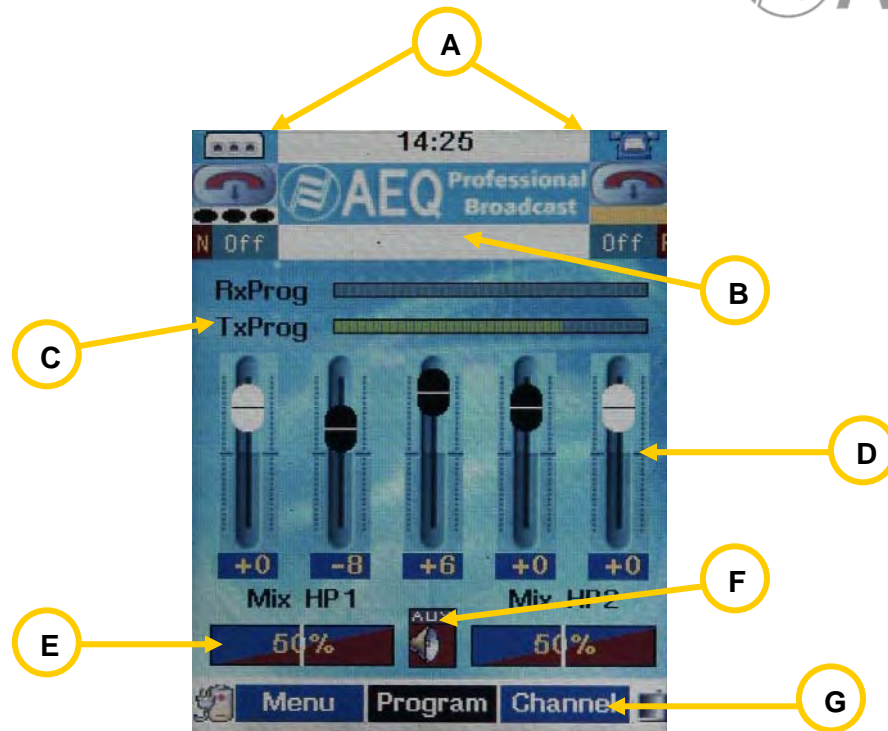
2.3.1.4. Multifunction display screen.

4

Shows the configuration and operating menus, information on the connection and audio VU meters, which can measure input and output audio signals.

The AEQ PHOENIX MOBILE includes a 240x320-pixel color TFT screen that displays the internal codec menu, which is accessed by using the set of keys located under display.

Through this menu you can access, by pressing the navigation arrow keys, the various configuration submenus: press the up and down arrows to scroll through the menus, and the left and right arrows to change the values.









Start-up or idle screen detail

The start-up screen is the first to be displayed, by default, when the unit is started, and will continue to be displayed in the idle state unless you perform operations.

1.- Communication channel section. A




The screen is divided into two parts: the left-hand segment corresponds to the Program channel, and the right-hand part shows the Coordination channel.

In the upper part of the module one of the following images is shown:

-  if that communication channel is using the Ethernet port for communication.
-  if the transmission is going out over a POTS (PGA-01) communication module.
-  if the transmission is going out over an ISDN (PGA-03) communication module.
-  if the transmission is going out over a mobile phone connected to USB port.
-  if through that mobile phone a **3G** communication is established.
-  if there is no module in the selected slot.

The assignment can be modified in **Menu >> Modules Config. >> Ch. Configuration**. For each one of those images, the **background** can be **blue** (module status OK), **orange or red** (no connection) or **green** (established communication). The background of **3G** icon can be **yellow** (while negotiating with your phone provider the communication parameters and IP address), **red** (while PHOENIX MOBILE is registering into the SIP server) or **blue** (once the process, that may take up to 10 seconds, is completed).

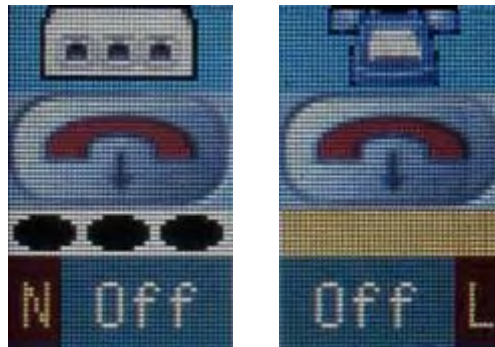
Beneath these symbols an icon representing a telephone handset is displayed:

-  indicates that it's on-hook and there is no communication established.
-  indicates an incoming call.
-  indicates that there is a communication established.

Just below, there is a space whose function is to indicate the channel that is currently selected (**three black points** are visible on the screen), and whose name can be displayed in the central black space in the lower part of the screen: white text for **Program** and yellow for **Coord**.

In the lower part of this section, both in Program and in Coordination, you can read:

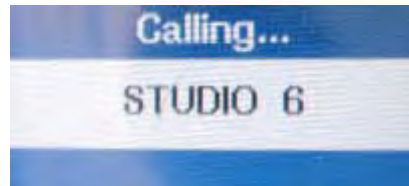
- **On** or **Off**: Indicates whether the connection with the communications network is established or not.
- **N, L, R** or **U**: The communications channel (Prog or Coord.) is established through the Ethernet interface (N), the left additional comm. module (L), the right additional comm. module (R) or the USB interface (U).



Detail of communication channel section

2.- Call status information section. **B**

In the idle state, only the time is displayed. When there is an outgoing or incoming call, the called or calling telephone number is displayed and, once the call is accepted, the information on the progress of the negotiation / synchronization of the call is shown.



Detail of the section showing information during a call

3.- VU meter section. **C**

Two mono VU meters, and the name of the output or input channel whose level they represent. The assignment can be modified in **Menu >> Applications >> VU Meters**.

4.- Section showing input / output signal level. **D**

Gives real-time graphic display of the level of the signal selected for the inputs / outputs of the unit. From left to right, the inputs / outputs identified are HP1, MIC1, MIC/LIN, MIC2 and HP2. The ranges are: for HP1 and HP2, from +12 to <-78, and for MIC1, MIC/LIN and MIC2, from +12 to <-62.

5.- Headphone balance section. **E**

Exactly the same for HP1 and HP2: by pressing on the encoder corresponding to its output on the control surface, you will disable the operation on the signal level (as well as the movement of the associated fader displayed, changing to a darker color than what is normally shown to reflect this action), and then the desired balance selection for these headset outputs will be displayed in real time.

6.- Auxiliary Output active indicator. F



If the loudspeaker icon is visible on the start-up screen, it means that the Auxiliary output is active; if the icon is not displayed, the output is off. This can be modified in **Menu >> Audio settings >> Output >> Aux Out.**

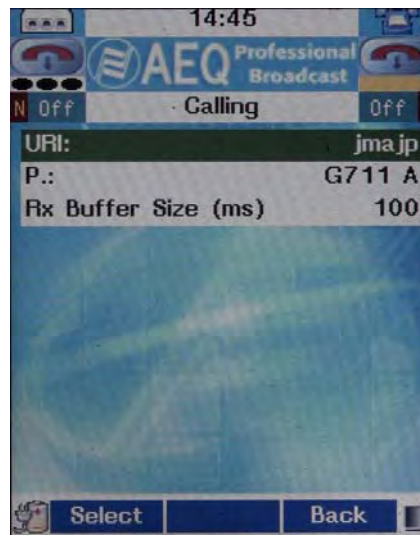
7.- Lower section. G

Three icons: "Menu", "Program" or "Coord." and "Channel".

- **"Menu"**: By pressing the left navigation key you will access the general internal menu for the unit.
- **"Program"** or **"Coord."**: Indication of the communication channel selected.
- **"Channel"**: Press the right navigation key to switch between "Program" and "Coord".

From the initial screen, pressing the center navigation button will display the last number or URI called (ready to reconnect by simply pressing the green button on the control surface), the communications profile to be used (see the "Link Profile" section in this manual for more information), and the maximum size of the reception buffer (only for IP and GSM communications) or the maximum baud rate (only for POTS module when AAC-LC or HE-AAC encoding mode is selected). All the fields here can be modified with the "Select" option.

This screen is also accessible from the fast-access **"CHN"** button on the control surface.



CHN detail

2.3.2. Alphanumeric keypad.



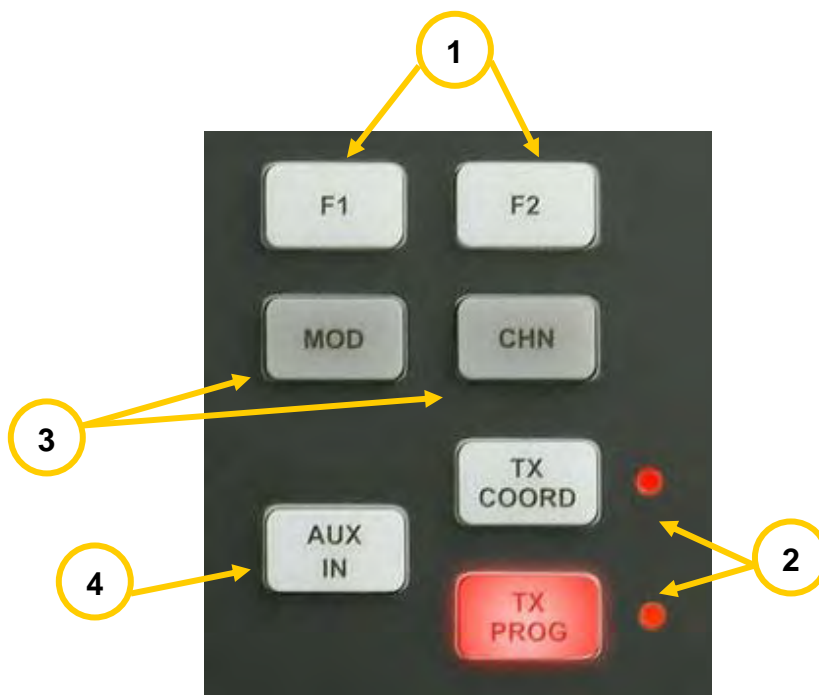
This is a normal alphanumeric keypad that allows you to key in a number when you want to make a call using POTS, ISDN or USB-GSM interface or numbers and text when you want to make a call using IP or USB-3G interface (or when you are looking for or editing phone book entries).

To change between different letters of the alphabet associated with the same key, press the key repeatedly until desired letter appears, just as you would with a cell phone. The navigator up and down arrows will enable you to select upper and lower case letters and numbers. The left arrow key acts as a backspace key and the right one as a space bar.

Enables the speed dialing function by pressing an entry number stored in the phone book plus “#” (see section 3.2) . It also allows DTMF (multifrequency) dialing tones to be generated.

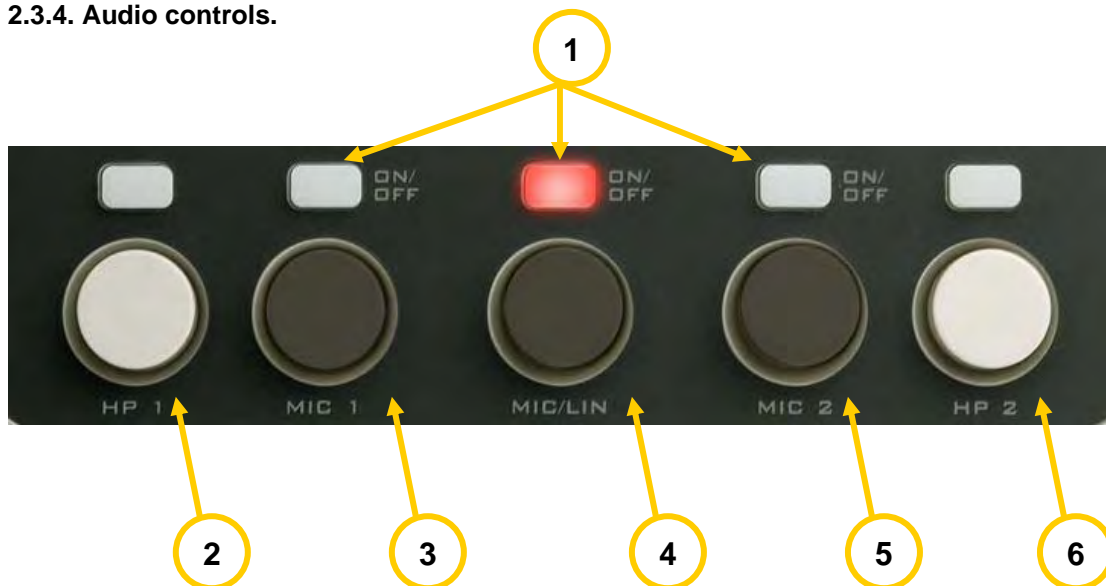
The keys backlighting blinks to notify the user of an incoming call when the buzzer (audible notification of calls) is OFF.

2.3.3. Function and routing keys and communication status LEDs.



- 1 **F1** and **F2** function keys: These keys give you fast access to the Phonebook and the List of available SAP transmissions, respectively.
- 2 **TX COORD** and **TX PROG** keys, and status **LEDs**: These keys let you activate the two communication circuits: Program and Coordination. You can send input signals to these communication circuits in the **Audio settings >> Routing** menu (for additional information, please consult section 3.3.1 in this manual). The two status LEDs indicate the status of the communication channels associated with the communication circuits:
 - **Red**: if there is a problem in the channel that prevents communication from being established.
 - **Orange**: the communication is established but not synchronized.
 - **Green**: the communication is established and synchronized.
- 3 **MOD** and **CHN** keys: Direct access to the configuration of communication modules menu (MOD) and call configuration menu (CHN).
- 4 **AUX IN** key: Auxiliary input activation key: activates or deactivates the sending of audio from the auxiliary input to the internal mixer in the unit.

2.3.4. Audio controls.



- 1 **Channel activation keys**: enable you to send audio from the “MIC 1”, “MIC 2” or “MIC/LIN” inputs to the internal mixer in the unit. When the LED is lighted, it indicates ON.
- 2 **Headphone Encoder 1**: turn this control to select the level of audio available in the Headphone 1 output (“HP 1”). By pressing the button, you will change the control into an audio listening balance potentiometer. This selection is shown on the screen. When you press again, the control returns to its normal functionality. The key placed above the encoder allows you to active/deactivate the sending of signal to the Headphone 1 output (“HP 1”).
- 3 **Microphone Encoder 1**: by turning this control, you will modify the level of the input audio coming from microphone 1 (“MIC 1”). The level is displayed on the screen as variations on a fader. When you press the button, you will access the configuration menu for this input (see section 3.3.2.1 of this manual). When you press again, the control returns to its normal functionality.

- 4** **Microphone / line Encoder:** by turning this control, you will modify the level of the input audio coming from microphone / line input (“MIC / LINE”). The modification is displayed on the screen as variations on a fader. When you press the button, you will access the configuration menu for this input (see section 3.3.2.2 of this manual). When you press again, the control returns to its normal functionality.
- 5** **Microphone Encoder 2:** by turning this control, you will modify the level of the input audio coming from microphone 2 (“MIC 2”). The modification is displayed on the screen as variations on a fader. When you press the button, you will access the configuration menu for this input (see section 3.3.2.1 of this manual). When you press again, the control returns to its normal functionality.
- 6** **Headphone Encoder 2:** turn this control to select the level of audio available in the Headphone 2 output (“HP 2”). By pressing the button, you will change the control into an audio listening balance potentiometer. This selection is shown on the screen. When you press again, the control returns to its normal functionality. The key placed above the encoder allows you to active/deactive the sending of signal to the Headphone 2 output (“HP 2”).

2.4. Ethernet Interface.

PHOENIX MOBILE is a portable encoder over IP, fully compatible with the **EBU-TECH 3326** technical specification formulated by the **EBU N/ACIP** working group, which was developed to guarantee compatibility between different manufacturers in applications for contributing professional quality audio over IP networks. For further information, consult:

http://www.ebu.ch/CMSimages/en/tec_doc_t3326-2008_tcm6-54427.pdf

The connector used is an **RJ45** for 10 Base T interface.

2.5. PGA Interfaces.

In addition to the Ethernet connection, the device has two extra slots for inserting additional communication modules, such as POTS or ISDN. This modular design makes the PHOENIX MOBILE easy to update in the future with new communication modules responding to the needs whose satisfaction the market may demand at any time.

Besides the Program signal and its corresponding return, the PHOENIX MOBILE can simultaneously provide a channel for redundancy or Coordination purposes by making use of the additional communication modules.

The PHOENIX MOBILE is designed to ensure interoperability with existing and future communications equipment developed by AEQ and other manufacturers, thanks to the large number of coding algorithms it implements, and to the possibility of easily bringing them up to date as the market evolves.

2.5.1. PGA-01 communications module: POTS.

POTS is the acronym for **Plain Old Telephone Service** (also known as traditional telephone service), which refers to the way in which analog (or conventional) telephone service is offered over copper wires. In Spanish, this is called RTB (Red Telefónica Básica, Basic telephone network).



Detail of POTS module

The POTS communications module enables you to use different working modes, depending on your needs:

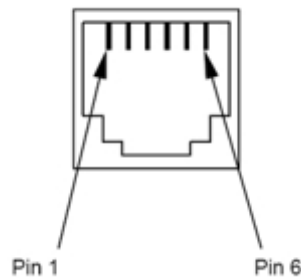
- POTS modem
- Telephone hybrid
 - Normal function
 - Frequency extension

For transmission over traditional telephone networks, using a bandwidth ranging from 300 to 3400Hz in the normal mode.

The frequency extension function allows you to transmit the 50 to 300 Hz frequency range over the telephone line. To accomplish this, the signal sent to a 250-Hz frequency shift, improving the quality of the received signal in its lower frequencies at the expense of the higher range. The transmitted bandwidth ranges, then, from 50 to 3150 Hz. The 250 Hz lost in the highest frequencies are not very significant, due to the logarithmic nature of audio frequency response.

The frequency shift is accomplished through the encoding of the audio signal before it is sent to the telephone line. On the receiving end the signal is decoded--that is, the reverse frequency shift is performed, an operation that must be carried out by suitable equipment. The decoded signal falls within the original range (50-3150 Hz) without having suffered any sort of interference. In this way, greater depth and clarity is achieved in the voice signal, even in those communications that take place under the worst conditions.

The physical interface is a standard **RJ11** connector.



Detail of RJ11 connector

2.5.2. PGA-03 communications module: ISDN.

According to the UIT-T we can define Integrated Services Digital Network (ISDN) as a network that proceeds for evolution of the Digital Integrated Network and that facilitates end-to-end digital connections to provide a wide range of services, both of voice and of other types, and to that the users accede across a set of normalized interfaces.

It is possible to say then that it is a network that comes for evolution of the telephonic existing network, which on having offered digital connections of end to end allows the integration of multitude of services using an unique access, independently from the nature of the information to transmitting and of the terminal equipment that generates it.



Detail of ISDN module

The connector used is an standard **RJ45** connector for S interface and an standard **RJ11** connector for U interface.

2.5.3. Other PGA communication modules.

In this area, depending on market needs, when the time comes, other communication interfaces will be marketed for GSM, 3G, Wifi/Wimax, Bluetooth, etc.

2.6. USB Interface.

The USB B port of PHOENIX MOBILE allows you to connect a mobile phone (by means of mobile phone data cable and an USB B to UBS A adapter) in order to establish a **GSM** or **3G** communication. When plugging a mobile phone, it will be automatically recognized by PHOENIX MOBILE. In order to make the connection easier, AEQ recommends that you use a mobile phone that use an standard USB data cable.



USB adapter and 3G mobile phone with data cable



Connection of mobile phone to PHOENIX MOBILE

IMPORTANT NOTE: You should be aware that, during data transmission, mobile phone battery consumption rises considerably.

2.7. Batteries pack.

The PHOENIX MOBILE is supplied by default with a battery charger / power supply and, as an option, can include a batteries pack of the Li-Ion type (3500 or 4000mAh / 7.4V), which will power the codec for an average of more than 2.5 hours in normal operating mode, with easy removal as an additional feature, and whose machining includes an integrated carrying handle.



Details of the unit with batteries pack during transport (left), during operation (centre) and batteries pack (right)

The status and operation of the batteries pack associated with the unit, if such a pack is connected, is displayed on the main screen by means of two icons shown in the lower left- and right-hand corners.

- The icon in the lower left-hand corner indicates whether the unit is being powered:
 - by the batteries pack
 - or directly by mains power through the power supply/charger supplied

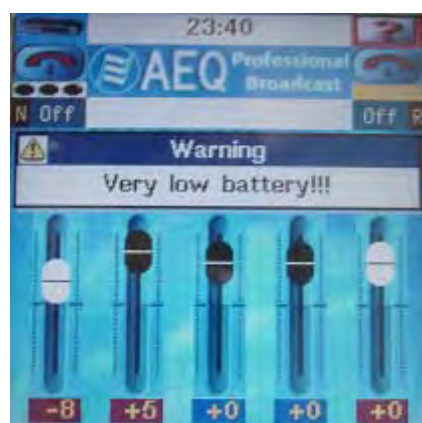


The second option happens when the batteries pack is connected to the unit and, at the same time, to mains power through the power supply/charger supplied: in that case, in addition to power the unit, the power supply/charger allows the simultaneous batteries pack charging.

- The icon in the lower right-hand corner indicates the charge level in the batteries pack.



For more information, please consult section 3.7.2 in this manual



Very low battery level warning

WARNING:

1. Never tamper with the batteries.
2. Always use original AEQ power supplies and, in case of failure, send the malfunctioning part to AEQ or one of its authorized agents for repair or replacement.

3. INTERNAL MENU DESCRIPTION.

From the start-up screen, press the left navigation button associated with the Menu option that is shown in the lower left-hand portion of the display, to enter the main menu of the unit.

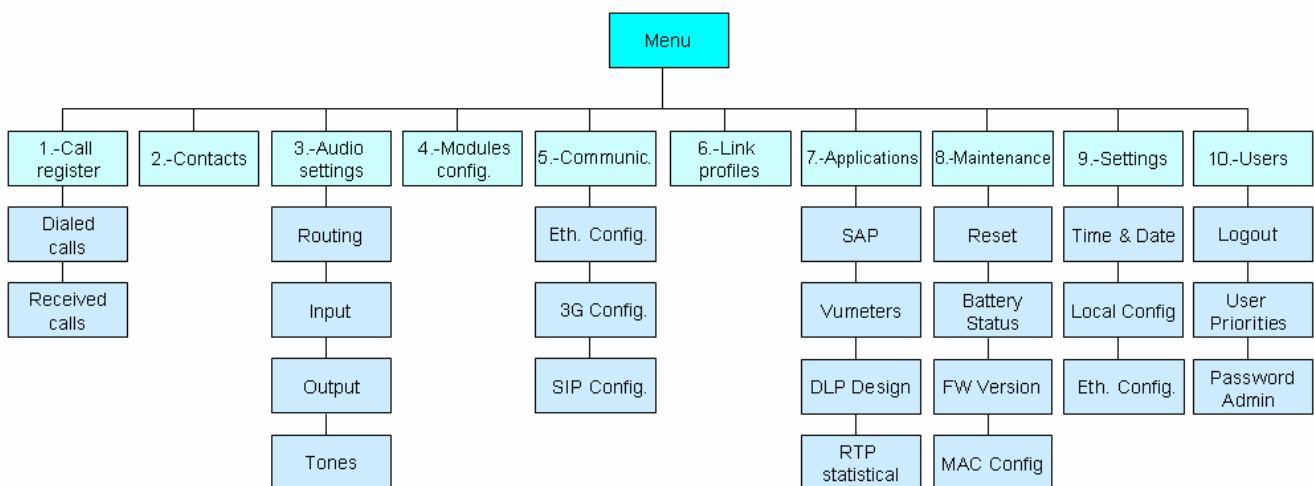


Menu access



Internal menu diagram

The internal diagram of this main menu is detailed in the following paragraphs.



Main menu organization diagram

From this menu you can configure all of the operational parameters of the unit.

Each one of the fields that comprise it contains, in turn, more internal fields that are specific to the functionality in question, and which will be described in detail in the relevant section of this manual.

- 1) **Call register:** Information regarding calls made or received.
- 2) **Contacts:** Phone book listing contacts.
- 3) **Audio settings:** Configuration of routing and mixing of the signals inside the unit, as well as configuration of the inputs and outputs in the equipment.
- 4) **Modules Config.:** Configuration of communication modules.
- 5) **Communications:** Configuration fo communication interfaces.
- 6) **Link Profiles:** List of communication profiles, with the encoding algorithms available in each case.
- 7) **Applications:** Advanced functions.
- 8) **Maintenance:** Codec internal maintenance menu.
- 9) **Settings:** Internal configuration of the unit.
- 10) **Users:** Users change and configuration.

3.1. Call Register.

Menu in which the information regarding the last ten calls made or received (accepted and rejected) through any of its communication interfaces is stored in the unit. These data are stored in order from the most recent (placed on top of the list) to the oldest (placed at the bottom).

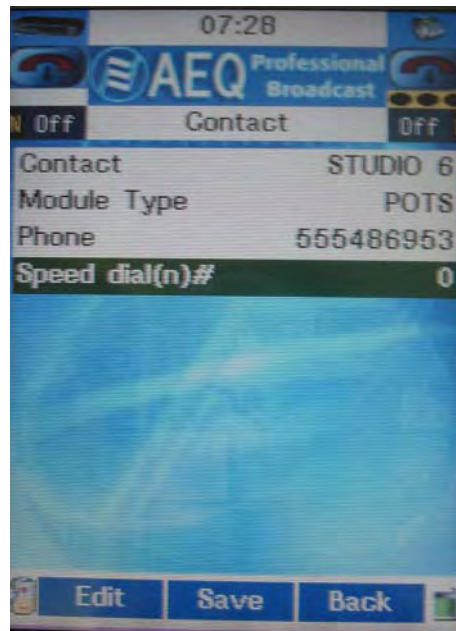
On each one of these entries shown on the screen you can press the **Select** option and display detailed information regarding the call in question:

- Telephone number or URI (Universal Resource Identifier for IP communications), depending on the interface over which the communication was established.
- Name of the contact, if this entry is contained in the internal phone book of the equipment. If the contact name is not in the internal phone book, it is identified as "Unknown".
- Channel used for the communication: "Program" or "Coordination".
- Communication module over which the connection was established: IP or Phone (POTS, ISDN or GSM).
- Information about date, time and length of the communication.

3.2. Contacts.

Internal phone book stored in the codec, organized alphabetically, with a capacity for 255 entries. For each of the entries, the following details can be displayed:

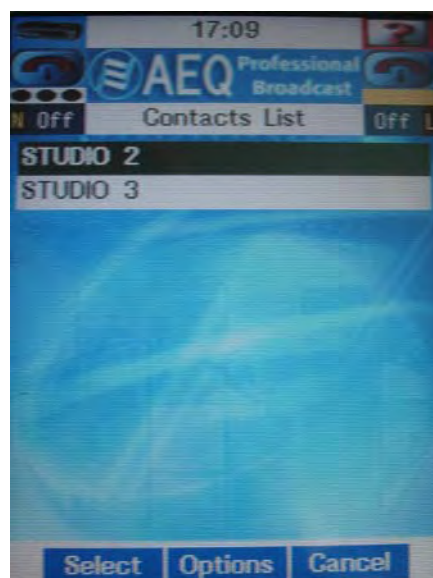
- Name of the contact, to be filled in by using the alphanumeric keypad. The navigator up and down arrows will enable you to select upper and lower case letters and numbers. The left arrow key acts as a backspace key, and the right one as a space bar.
- Communication interface over which you wish to establish communication: IP or PHONE (POTS, ISDN or GSM).
- Associated number or URI (depending on whether the communication module to be used is POTS, ISDN, GSM or IP).
- Speed dialing number. It's the number associated to that entry of the phone book. When you press it followed by "#" the speed dialing mode is activated.



Details of a phone book entry

On each of the entries, pressing the center button associated with the **Options** option will enable you to:

- Create a new entry: **Menu >> Contacts >> Options >> Add**. Fill in the contact's name, the communication module, URI or telephone number and speed dialing number.
- Call: **Menu >> Contacts >> Options >> Call**. This action launches an immediate call to the selected contact.
- Edit an entry: **Menu >> Contacts >> Edit**. You can edit any or all of the four available fields.
- Eliminate an entry: **menu >> Contacts >> Options >> Delete**. This operation eliminates the contact you have selected. The system will ask for your confirmation.
- Completely empty the phone book: **Menu >> Contacts >> Options >> Delete All**. This action eliminates all the contacts stored in the phone book. The system will ask for your confirmation.

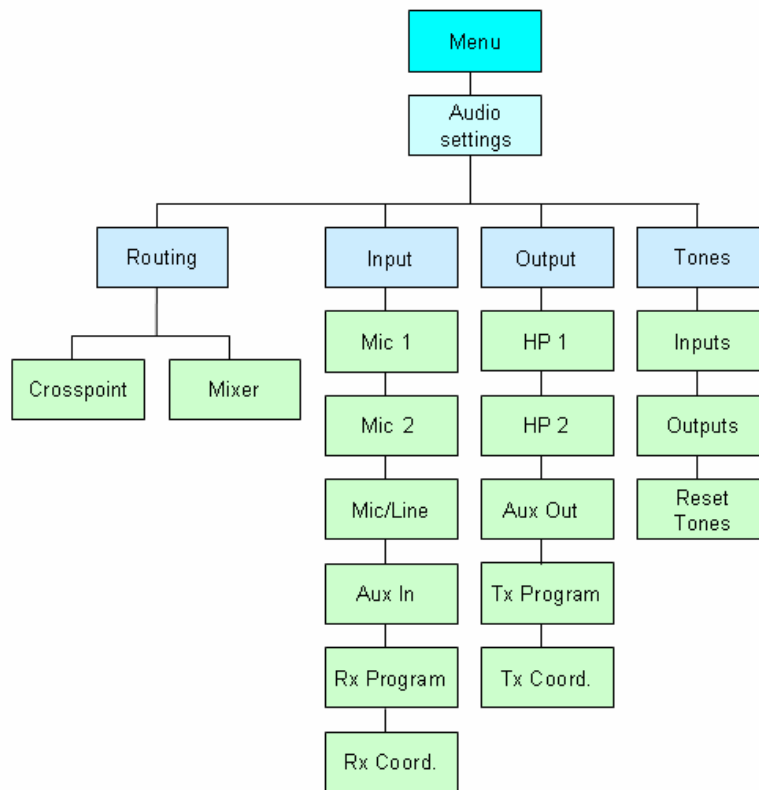


Phonebook detail

3.3. Audio settings.

The PHOENIX MOBILE is a complete router and digital audio mixer. Although its inputs and outputs (MIC, HP and AUX) are analog, the first and the last step is a 24-bit A/D and D/A converter with sampling frequency of 48KHz.

Menu >> Audio settings is the section of the menu where all the routing and mixing functions of the unit are managed, and where the inputs and outputs available in the equipment are assigned and configured.



Audio settings menu diagram

3.3.1. Routing.

The access to this menu is:

Menu >> Audio settings >> Routing.

This menu allows you to configure the routing and mixing of the input and output signals from and to the different communication channels, in both the normal mode (mono) and advanced CUSTOM mode in which the headphones are treated as stereo outputs; in the CUSTOM mode you can make the sends independent from each other within the same headphone to the right and left channels.

It is divided into two areas called “Crosspoint” and “Mixer”. The “Crosspoint” area manages the sends (routing) between input and output channels in the unit, and the communication channel (“Program” and “Coord.”) sends and returns; and the “Mixer” area configures the send levels associated with each crosspoint you establish.

3.3.1.1. Crosspoint.

When you select this option, a list of the outputs which routing can be configured is shown:

- Tx Program
- Tx Coord.
- HP 1
- HP 2
- Aux Out

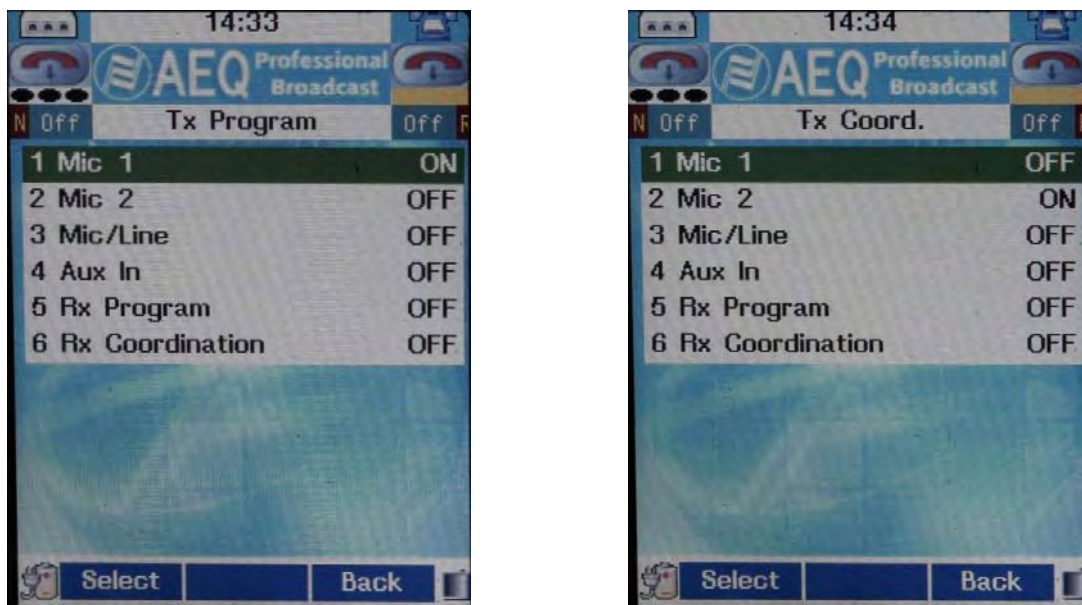
If any of the headphones is defined as “**Custom**”, it will appear with two items, one for the left channel and the other one for the right channel:

- HP 1 Left
- HP 1 Right
- HP 2 Left
- HP 2 Right

By selecting one of this outputs, a list of the internal matrix inputs that can be routed to that output is shown. The internal matrix inputs that are available for each one of the outputs are:

- Mic 1
- Mic 2
- Mic/Line
- Aux In
- Rx Program
- Rx Coordination

The send or crosspoint is established when ON and disabled when OFF. You can make this modification by pressing the key associated with the **Select** option in the screen.



Example of established crosspoints: MIC1 input to TX PROGRAM output and MIC2 input to TX COORD. output

A routing example in the basic option (both headphones configured as “**Mono**”) is shown next:

EXAMPLE OF MONO COMMENTARY ROUTING

		6 INPUTS					
		1	2	3	4	5	6
		M I C	M I C	M I C / L I N E	A U X	P R O G R X	C O O R D R X
O U T P U T S	1	HP1	HP1L				
	2	HP1	HP1R				
	3	HP2	HP2L				
	4	HP2	HP2R				
	5	AUX OUT					
	6	PROG TX					
	7	COORD TX					

Example of HP routing in normal mode

In case you have defined both headphones as “**Custom**” (**Menu >> Audio settings >> Output >> HP 1 and HP 2**), you could configure the routing, for instance, the following way:

EXAMPLE OF STEREO CHANNEL ROUTING

		6 INPUTS					
		1	2	3	4	5	6
		M I C	M I C	M I C / L I N E	A U X	P R O G R X	C O O R D R X
O U T P U T S	1	HP1	HP1L				
	2	HP1	HP1R				
	3	HP2	HP2L				
	4	HP2	HP2R				
	5	AUX OUT					
	6	PROG TX					
	7	COORD TX					

Example of HP routing in custom-stereo mode

The PHOENIX MOBILE units come from the factory configured with the following routing structure:

- MIC1 -> Tx Program.
- MIC2 -> Tx Coord.
- Rx Program -> HP1 (in mono mode).
- Rx Coordination -> HP2 (in mono mode).

3.3.1.2. Mixer.

In the Mixer area you can configure the send level of each input to each of the following outputs:

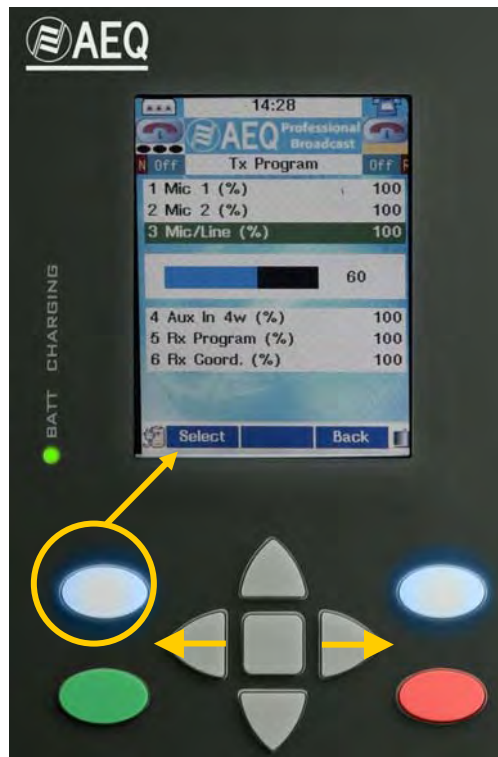
- Tx Program
- Tx Coord.
- HP 1
- HP 2
- Aux Out

If any of the headphones is defined as “**Custom**”, it will appear with two items, one for the left channel and the other one for the right channel.

For each one of these outputs, you can select the send level of the following input signals:

- Mic 1
- Mic 2
- Mic/Line
- Aux In
- Rx Program
- Rx Coord.

By pressing the button associated with the **Select** option, you can select the send value from a range starting with a maximum represented by the value 100 (to send the signal in question with the exact same level set in the input) and ranging down to 0 (to send the signal completely muted). The intermediate values are 10-unit jumps which are selected by pressing the left and right arrows of the navigation buttons.



Detail of mixer and button associated with the Select option

NOTE: Regardless of the level selected in this section, the routing will not take effect until the corresponding crosspoint is activated in the section **Menu >> Audio settings >> Routing >> Crosspoint.**

The PHOENIX MOBILE units leave the factory configured with all the mixing factors at 100%.

3.3.2. Input.

This menu allows to configure and activate each one of the inputs of the unit (“Mic 1”, “Mic 2”, “Mic/Line”, “Aux In”, “Rx Program” and “Rx Coord.”).

The access to this menu is:

Menu >> Audio settings >> Input.



Input menu

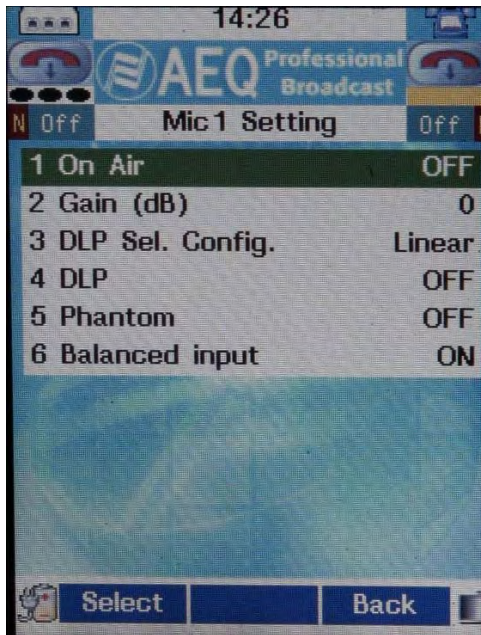
3.3.2.1. Mic 1 and Mic 2.

The MIC1 and MIC2 microphone inputs allow you to make the following configurations:

- **On Air:** Lets you activate (ON) or deactivate (OFF) the channel. When this parameter is ON, the button corresponding to this input activation on the control surface will light up. When it is OFF, the button will remain unlighted.
- **Gain (dB):** Allows you to modify the input gain between +6dB and -37dB.
- **DLP Sel. Config.:** Allows you to select the configuration for DLP, from the list of options previously configured as available. DLP configuration is explained in annexe D.
- **DLP:** Lets you active (ON) or deactivate (OFF) the audio processing that allows to improve the quality of input audio. This processing enables for example a compression, a limiter and a noise reducer to be applied.
- **Phantom:** Allows you to activate (ON) Phantom feed (12V DC) for use in case an electrostatic microphone is connected to that input.
- **Balanced input:** Activating (or deactivating) this parameter enables a better adaptation of the input module for balanced (or unbalanced) signals. In professional applications, we recommend always to work with balanced signals.

3.3.2.2. Mic/Line.

For the Mic/Line input, the options are the same that for MIC1 and MIC2, except that the “Phantom” option is not available, and is replaced by the option “**Mode**” that allows you to select between two operating modes: **microphone input level** or **line input level**. In case that line level is selected, the input gain can be modified between 0dB and +53dB.



Detail of an input channel and the activation button on the control surface

3.3.2.3. Aux In.

For the “Aux In” input, the available options are channel activation (with lighting of the button corresponding to this input on the control surface), gain control and DLP options.

3.3.2.4. Rx Program and Rx Coord.

For the Program and Coordination return inputs, the only available options are channel activation (letting that signal pass to the internal digital mixing module inside the unit) and gain control (between 0dB and -80dB).

3.3.2.5. Default configuration.

The PHOENIX MOBILE units leave the factory configured with the following parameters:

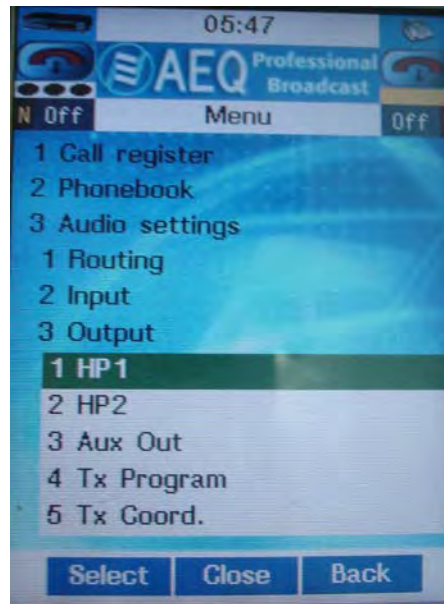
- Offset of nominal gain adjustments at 0dB (Gain in “Input” menu).
- Input gains at the nominal level of 0dB (encoders on control surface).

3.3.3. Output.

This menu allows you to configure and activate each one of the outputs of the unit. “HP 1”, “HP 2”, “Aux Out”, “Tx Program” and “Tx Coord.”.

The access to this menu is:

Menu >> Audio settings >> Output.



Output menu

3.3.3.1. HP1 and HP2.

On the HP1 and HP2 outputs, you can activate the channel (which will cause the button corresponding to that output to light up on the control surface) and configure the working mode as **"Mono"** or **"Custom"** (for more information, please consult section 3.3.1.1 in this manual).

3.3.3.2. Aux Out.

On the "Aux Out" channel, you can activate the channel (which will make an AUX loudspeaker icon appear on the start-up screen as a graphic indication) and set the gain between 0dB and -80dB.

3.3.3.3. Tx Program and Tx Coord.

You can activate the "Tx Program" and "Tx Coord." channels (an action that will cause the button corresponding to that output to light up on the control surface as an additional and immediate indication) and set the gain between 0dB and -80dB.

3.3.3.4. Default configuration.

The PHOENIX MOBILE units leave the factory configured with the following parameters:

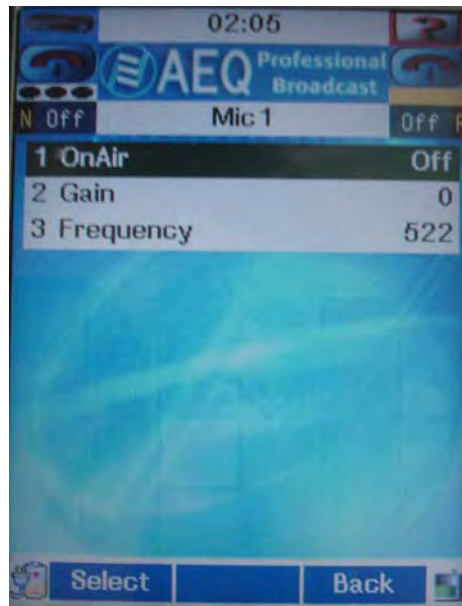
- Offset of nominal gain adjustments at 0dB (Gain in "Output" menu).
- Output gains at the nominal level of 0dB (encoders on control surface).

3.3.4. Tones.

This section allows you to use the unit as a tone generator for all the inputs and outputs available in the equipment.

In this section the headphones are always treated as stereo lines, where the sends to the left and right channels can be made independently.

For each channel (regardless of whether it is input or output) in **Menu >> Audio settings >> Tones >> Inputs** or **Output** you can adjust the level (between +12dB and -78dB in 6dB steps) and frequency (in 87Hz steps) of the tone to be generated, but your adjustments will have no effect on the tone until you activate the selection ("OnAir" ON).



Tones generation menu

Finally, there is a **“Reset Tones”** option which enables you simultaneously and definitively to turn off all the tones activated. This option is executed by shifting the field **Menu >> Audio settings >> Tones >> Reset Tones** from OFF to ON: by pressing Select and selecting ON, the tones will immediately cease and that field will return to OFF value.

3.4. Modules Config.

Menu for configuring the communication modules.

The access is **Menu >> Modules Config.**; it can also be accessed by pressing the quick access button **“MOD”** in the control surface.



Modules Config. menu

3.4.1. Associating Communication Interfaces with Channels.

“Ch. Configuration” section enables you to select the communication interface that will be used for **Program** (“Ch. Program”) and **Coordination** (“Ch. Coord.”).

Both channels are full-duplex, but Program is reserved for an encoded channel which will consequently be a high quality channel, while Coordination is a lower quality channel.

IMPORTANT NOTE:

- When associated to **Coordination** channel, only G711 A-Law, G711 μ -Law, G711 A-Law Extended and G711 μ -Law Extended algorithms will be available over **ISDN interface**, only G711 A-Law and G711 μ -Law over **Ethernet interface** and only Phone over **POTS interface**. When **USB interface** is associated to **Coordination** channel, only a **3G** communication with G711 A-Law and G711 μ -Law algorithms can be established; no **GSM** communication can be established over that channel.
- It's not possible to associate at the same time, in Program and Coordination or vice versa, the **USB** interface with **3G** mode active and the **Ethernet** interface. An error message is shown on the screen (“Modules not compatible” or “Setting may not be changed”) when you try to associate one of this modules when the other one is active.

In both cases, the options are:

- **Ethernet:** Basic communication module for IP communication.
- **Right Slot:** Additional communication module 1.
- **Left Slot:** Additional communication module 2.
- **USB:** Additional communication module for GSM or 3G communication.

By default, PHOENIX MOBILE is supplied from the factory with Ethernet interface associated with Program and right slot associated with Coord.

3.4.2. Optional Communication Interfaces Configuration.

“Removable Slot Config” section allows you to configure each of the insertable optional communication modules and interfaces.

3.4.2.1. [R].

This section indicates the class (POTS, ISDN, etc.) of the right additional communication module.

If the slot is empty, “Empty” will appear and “Config” option is not available.

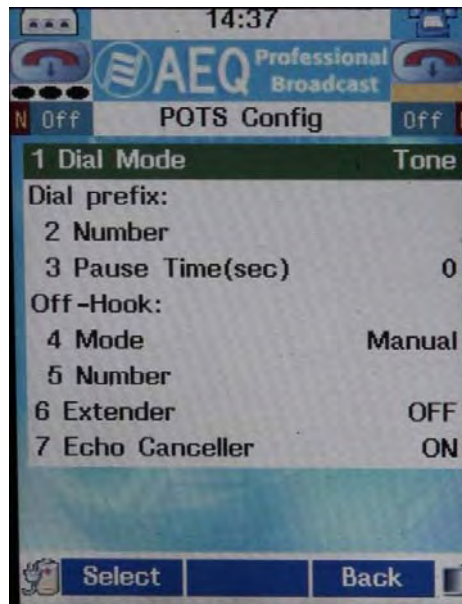
Pressing “Config” will take you to the advanced configuration mode.

3.4.2.1.1. Configuration of the slot for the POTS module.

If an additional PGA-01 model POTS communication module is inserted into the slot, the following configuration options will appear:

- **Dial Mode:** Allows you to select the dialing mode between tones and pulses.
- **Dial prefix Number:** Allows you to add a telephone code, such as the number for seizing a line when accessing the PSTN through a PBX.
- **Dial prefix Pause Time:** Allows you to configure waiting time before continuing to dial after seizing a line through a PBX, for example. Between 0 and 5 seconds.
- **Off-Hook Mode:** It can be set as automatic or manual.
- **Off-Hook Number:** Allows you to assign a telephone number to enable the POTS module to off-hook automatically only when calls from that number are received. When calls come in from other numbers, the unit reports their reception and behaves as it would be in manual answering mode, never rejecting any incoming call.

- **Extender:** Allows you to activate/deactivate the frequency extender.
- **Echo Canceller:** Allows you to activate/deactivate the echo canceller.



Detail of the configuration screen of the POTS PGA-01 additional communication module

3.4.2.1.1. Configuration of the slot for the ISDN module.

In case of inserting in the slot a module of additional communication model PGA-03 ISDN, two menus will appear: "ISDN Configuration" and "ISDN Parameters".



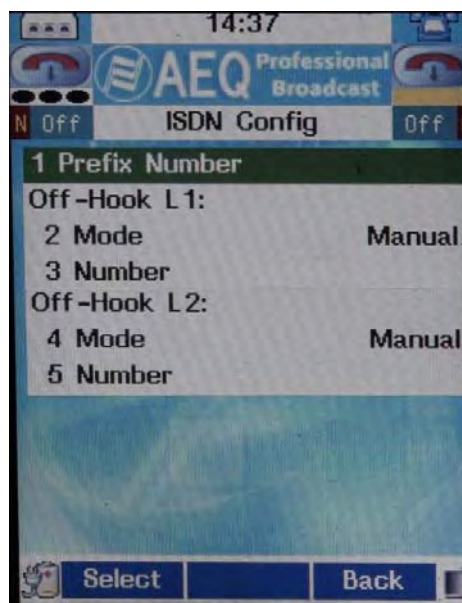
Detail of ISDN menu

"ISDN Configuration" menu allows to define the different options associated with the operation of the ISDN interface in the equipment, whereas "ISDN Parameters" menu allows to form the different technical parameters associated with The ISDN standard interface.

3.4.2.1.2.1. ISDN Configuration.

Inside this submenu the available options are the following ones:

- **Prefix Number:** Allows you to introduce regional prefix before the number associated with the ISDN interface.
- **Off-Hook L1 Mode:** Allows you to select the way of unhooked (manual or automatic) in the line 1 of the interface ISDN (first B channel).
- **Off-Hook L1 Number:** Allows you to associate a number with the option of automatic unhooked in line 1, only reacting to calls proceeding from the selected number. For the rest of calling numbers, the unit behaves as if the manual unhooked option is selected.
- **Off-Hook L2 Mode:** Allows you to select the way of unhooked (manual or automatic) in the line 2 of the interface ISDN (second B channel).
- **Off-Hook L2 Number:** Allows you to associate a number with the option of automatic unhooked in line 2, only reacting to calls proceeding from the selected number. For the rest of calling numbers, the unit behaves as if the manual unhooked option is selected.



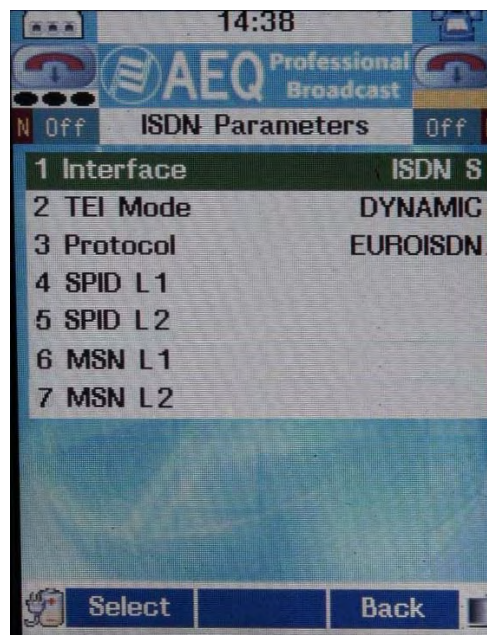
Detail of ISDN Config. submenu

3.4.2.1.2.2 ISDN Parameters

Inside this submenu the available options are the following ones:

- **Interface:** Allows you to select interface ISDN S or ISDN U, depending on the ISDN standard used.
 The interface U is the one that joins the telephone central office with the subscriber, is two wires as in the conventional analogical telephony and only connection admits point to point (only one device connected to the interface).
 The interface S/T is obtained from the interface U passing it for the TR1 placed in house of the subscriber, it is 4 wires and connection admits multipoint of up to 7 devices.
 In USA the companies do not give the TR1, being the user the one that must install it if wishes it. Normal situation will be that TR1 does not exist and therefore the access is in interface U. In short, in USA normal will be the access in interface U unless the subscriber has requested to the company or installed the same TR1 (the TR1 is normally a little box anchored to the wall with power cord, some luminous indicator, with external line of two wires input and RJ-45 output).
- **Tei Mode:** Terminal Equipment Identifier, allowing to select way of work between FIXED or DYNAMIC.

- Protocol:** Allows you to select **EUROISDN** or **National-1**.
 In USA in spite of several incompatible protocols between, thankfully exists a protocol standardized enough (named **National-1**) and that is in the habit of being available in all the plants. For more information, please consult annexe E.
 In Europe and other geographical zones there exists a protocol called **EURO ISDN**. There exist small differences between the EURO ISDN in the different countries. These differences do not affect the Phoenix Mobile.
- SPID L1:** Service Profile Identification Number for line 1.
 In case of the protocol National-1 besides the numbers of identification of line (DN's) it is necessary programme in the terminal the SPID's (Service Profile Identification Number) that are numbers that the carrier assigns associated to each of two channels of the line. Without the SPID's formed correctly the terminal will not work.
- SPID L2:** The same for the line 2 (second B channel).
- MSN L1:** Multiple Subscriber Number for line 1 (first B channel). It allows to form the number of the ISDN line contracted, in the physical interface of the equipment where it connects.
- MSN L2:** The same for the line 2 (second B channel).



Detail of ISDN Parameters submenu

In case of doubt, AEQ recommends to consult with your carrier or service provider, and see annexe E in this manual.

3.4.2.2. [L].

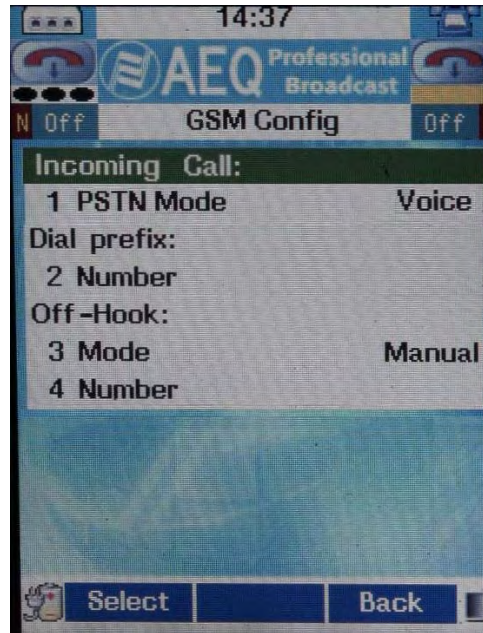
This section will allow you to configure the left additional communication module, in the same way as for the right slot, if the left module is installed.

3.4.2.3. [U].

This section will allow you to configure the additional communication module corresponding to USB interface. If the slot is empty, "Empty" will appear and "Config" option is not available.

When there is a mobile phone connected to the unit through USB port, "**Slot GSM**" indication will appear. Pressing "Config" will take you to the advanced configuration mode and the following configuration options will appear:

- **Incoming Call PSTN Mode:** This parameter must be configured as “Data” when the unit will receive a call coming from a POTS module and as “Voice” in any other case.
- **Dial prefix Number:** Allows you to add a telephone code.
- **Off-Hook Mode:** It can be set as automatic or manual.
- **Off-Hook Number:** Allows you to assign a telephone number to enable the POTS module to off-hook automatically only when calls from that number are received. When calls come in from other numbers, the unit reports their reception and behaves as it would be in manual answering mode, never rejecting any incoming call.



Detail of GSM Config. submenu

3.5. Communications.


Menu for configuring the communication interfaces.

The access is **Menu >> Communications.**

3.5.1. Ethernet Interface configuration.

Allows you to configure the IP parameters of the unit manually, when “DHCP” option is “OFF”, or automatically and transparent to the user (by using the DHCP protocol), when that option is “ON” (and the network where the unit is connected to allows to use that protocol). The parameters that will be configured are:

- **IP Local:** IP address of the communication interface.
- **Gateway:** Network gateway to which the equipment is connected.
- **Mask:** Mask associated with that address.

When DHCP option is activated, those three parameters are upgraded showing the values automatically assigned by the network. When you turn on the unit a question mark  appears next to the Ethernet interface symbol and it disappears as soon as the parameters have been assigned; if the network doesn't allow to use that protocol, after about 3 minutes the message “DHCP Error” appears. When DHCP option is active we recommend to use our **SIP server** as intermediary in the communications.

This configuration must be correctly done for the proper integration of the unit into the IP network that will support communication. If you have any doubts or questions, contact the Authorized Technical Services or with the System Administrator.

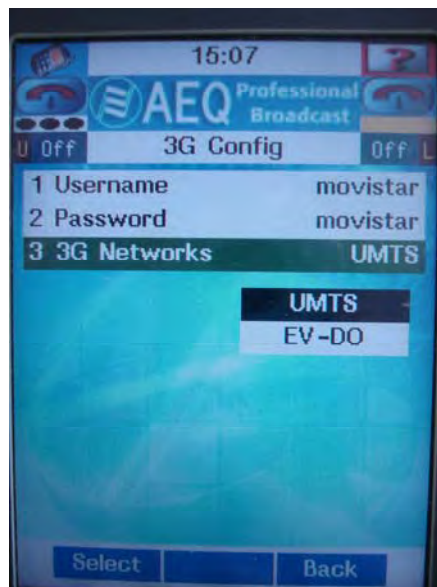


IP Configuration example

3.5.2. 3G Interface configuration.

Allows you to configure the 3G interface of the unit.

Typically, your mobile phone will be pre-configured with the operator's APN (Access Point Name) in order to function in 3G; all you need to configure in the PHOENIX MOBILE is the user name and password requested by your operator in order to have access to 3G connection and the 3G network type (UMTS or EV-DO). For further information on this subject, consult annexe E of this manual.



Detail of 3G Config submenu

3.5.3. SIP configuration.

This menu covers the configuration of the SIP, SAP and STUN parameters associated with the established IP communication, with four submenus called “Parameters”, “Proxy”, “SAP Configuration” and “Call Settings”.

3.5.3.1. Parameters.

This menu enables you to configure certain parameters associated with the SIP protocol implemented in the unit, as well as all of the STUN protocol parameters. For more information about both protocols, we recommend consulting annexe C.



Parameters submenu detail

- URI:** Identifier of the unit, if you are using the resources of an external SIP server; the format of this identifier will be of the **name@domain** type (for example, phoenix1@sip.aeq.es), where “name” is a mnemonic device associated with the codec, and “domain” identifies a proxy SIP server that will act as an intermediary in the communication, simplifying its establishment.
By default, PHOENIX MOBILE is supplied from the factory with a perfectly valid URI, ready to be used to establish communication.
IMPORTANT NOTE: the URI of the unit must not be changed if you want to use our **SIP server** as intermediary in the communications.
You also have the option of defining the identifier as name@IP-address. In this case, you will relinquish the use of the resources of the SIP server as an intermediary in the communication, and the connections will be of the direct IPaddress-to-IPaddress type.
- STUN Protocol Activate:** Allows you to activate/deactivate the STUN protocol use.
- STUN Protocol IP:** It shows the IP address of the STUN server, which will act as an intermediary in the communication, simplifying its establishment. If you have any doubts about this, consult your network administrator. By default, PHOENIX MOBILE will be supplied from the factory with a perfectly valid address, ready to be used to establish communication.
- STUN Protocol Port:** Identifier of the IP port over which you wish the STUN protocol to work, assuming it is activated. By default, PHOENIX MOBILE will be supplied from the factory with a perfectly valid port, ready to be used to establish communication.

3.5.3.2. Proxy.

This menu includes all configuration linked to the SIP Proxy, which will act as an intermediary in the communication, simplifying its establishment.

To simplify the operation of the unit, AEQ offers without aftercost the services of a proprietary SIP server, though 100% of the time can neither guarantee functioning nor make responsible for the disadvantages that it could suppose therefore the final user.



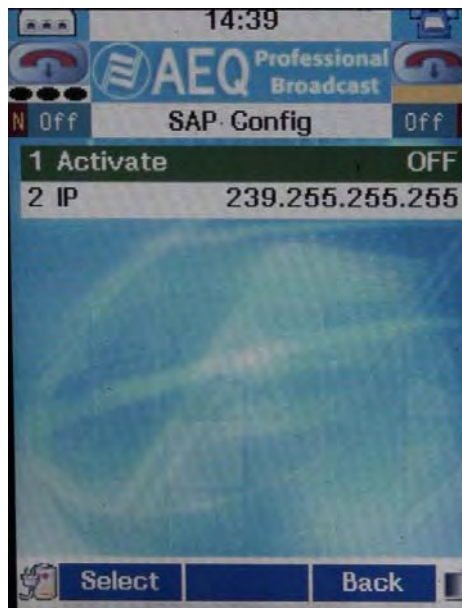
Detail of the Proxy submenu

- **Activate:** Allows you the activation/deactivation of the SIP Proxy function.
- **Address:** SIP Proxy server IP address. By default, PHOENIX MOBILE will be supplied from the factory with a perfectly valid address, ready to be used to establish communication.
- **Authenticate User:** Allows you to set the user name, required in the PHOENIX MOBILE authentication process undertaken with the SIP Proxy, during registration. By default, PHOENIX MOBILE will be supplied from the factory with a perfectly valid configuration, ready to be used to establish communication.
- **Authenticate Password:** Allows you to set the password, required in the PHOENIX MOBILE authentication process undertaken with the SIP Proxy, during registration. By default, PHOENIX MOBILE will be supplied from the factory with a valid password, ready to be used to establish communication.
- **Authenticate Realm.:** Allows you to set the realm under which the unit is registered in the SIP Proxy server. It will typically be the same as the SIP Proxy server domain, but by request and in coordination with the SIP Proxy server manager, it can be modified.
- **Register Domain:** SIP Proxy server identifier under which the PHOENIX MOBILE is registered. It must be the same as the domain described in the equipment URI.
- **Register Expires:** Every x minutes (an interval that is defined in this field), PHOENIX MOBILE will forward the information regarding its register in the SIP Proxy server so that the SIP server will not consider this equipment crashed and eliminate it as a consequence its register information. For example, the SIP Proxy server managed by AEQ has the maximum time set to five minutes, which means that the devices to be connected to this server must logically implement shorter "Register Expires" times.

IMPORTANT NOTE: these parameters must not be changed if you want to use our **SIP server** as intermediary in the communications.

3.5.3.3. SAP Config.

This menu includes the information relative to the SAP configuration. For further information about this protocol, we recommend consulting annexe C.

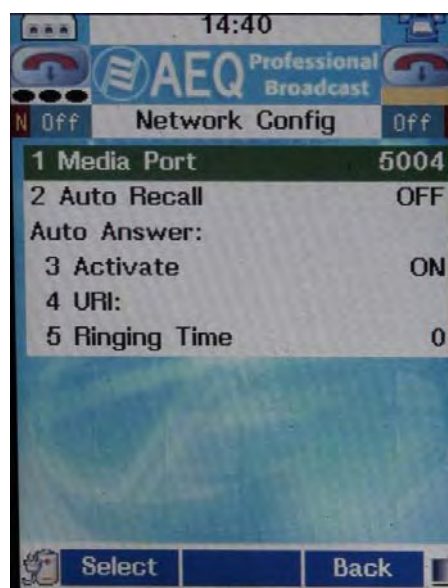


Detail of the SAP Config submenu

- **Activate:** Allows you the activation/deactivation of the SAP option in the equipment.
- **IP:** IP address of the SAP server, which will act as an intermediary in the communication, simplifying its establishment. If you have any doubts about this, consult your network administrator. By default, PHOENIX MOBILE will be supplied from the factory with a perfectly valid address, ready to be used to establish communication.

3.5.3.4. Call Settings.

This section allows you to establish the configuration of the IP communication interface in the equipment.



Slot IP menu

- **Media Port:** It defines the IP port selected as the input port to receive the audio information contained in the IP communication (by default 5004 ready to operate).
- **Auto Recall:** Allows you to activate/deactivate the recall option.
- **Auto Answer Activate:** Allows you to activate/deactivate the automatic answering option.
- **Auto Answer URI:** Lets you configure a default identifier, with respect to which the unit will automatically answer if it receives an incoming connection.
- **Auto Answer Ringing Time:** Allows you to set the length of the call notification, ranging from 0 seconds (without sound) to one minute, before automatic answering if this feature is enabled.

3.6. Link Profiles.

Menu displaying the list of communication profiles to be used in the IP communication (whether it is established through Ethernet interface or whether through USB-3G interface), with the encoding algorithms available in each case. The access is **Menu >> Link Profiles**.

PHOENIX MOBILE is an audio codec whose IP interface is absolutely compatible with the EBU-TECH 3326 standard formulated by the N/ACIP working group. This standard, originally conceived to ensure the interoperability of equipment made by different manufacturers in applications involving communications over IP networks, defines a series of algorithms as obligatory and others as optional. It is formulated so as to guarantee the interconnection between equipment items, through the obligatory algorithms, leaving the optional ones as an improvement to be used at each manufacturer's discretion. According to EBU-TECH 3326, the following modes are considered obligatory algorithms:

- ITU G.711
- ITU G.722
- ISO MPEG 1/2 Layer II

According to EBU-TECH 3326, the following modes are considered recommended algorithms:

- PCM
- ISO MPEG 1/2 Layer III (MP3)
- ITU G722.2 (AMR WB+)
- ISO MPEG 4 AAC
- APT-X
- MPEG 4 HE-AACv2
- DOLBY AC-3
- Manufacturers' proprietary modes

PHOENIX MOBILE includes all the obligatory algorithms and many of the recommended ones. We recommend using, for IP communication, encoding profiles that include at least one of the algorithms defined as obligatory, to provide a 100% guarantee of success in the negotiation and synchronization of communications.

In the rest of the communication interfaces, AEQ recommends ensuring that the encoding mode selected is compatible with the communication line.



List of profiles

By default the unit leaves the factory with the following profiles:

- AAC HE 22 Kbps mono 48KHz
- AAC HE 32 Kbps mono 48KHz
- AAC LC 32 Kbps
- AAC LC 64 Kbps
- AEQ LD EXT 128 Kbps
- G711
- G722
- HIGH QUALITY
- LOW DELAY
- MPEG LII 128 Kbps
- MPEG LII 64 Kbps

Each one of them contains only the encoding algorithm that coincides with the profile name, except the following ones, which are supplied by default as pre-defined lists.

- **AAC-LC 32 Kbps:** Comprehends the MPEG-4 AAC-LC 32 Kbps mono 48 KHz and AAC-LC 32 Kbps mono 24 KHz algorithms.
- **AAC-LC 64 Kbps:** Comprehends the MPEG-4 AAC-LC 64 Kbps mono 48 KHz and AAC-LC 64 Kbps mono 24 KHz algorithms.
- **G711:** Comprehends the ITU G.711 A-Law mono and ITU G.711 μ -Law mono algorithms.
- **HIGH QUALITY:** Comprehends the MPEG family of algorithms prioritized according to quality, the following way:
 - MPEG-1 LII 128 mono 48 KHz
 - MPEG-1 LII 128 mono 32 KHz
 - MPEG-2 LII 128 mono 24 KHz
 - MPEG-4 AAC-LC 64 Kbps mono 48 KHz
- **LOW DELAY:** Comprehends a set of algorithms, prioritizing according to the required delay, the following way:
 - AEQ LD-2 128 mono 32 KHz
 - ITU G.711 A-Law mono
 - ITU G.711 μ -Law mono
 - MPEG-2 LII 64 mono 16 KHz

- **MPEG LII 128 Kbps:** Comprehends the 128 Kbps MPEG LII family of algorithms, the following way:
 - MPEG-1 LII 128 mono 48 KHz
 - MPEG-1 LII 128 mono 32 KHz
 - MPEG-2 LII 128 mono 24 KHz

- **MPEG LII 64 Kbps:** Comprehends the 64 Kbps MPEG LII family of algorithms, the following way:
 - MPEG-1 LII 64 mono 48 KHz
 - MPEG-1 LII 64 mono 32 KHz
 - MPEG-2 LII 64 mono 24 KHz
 - MPEG-2 LII 64 mono 16 KHz

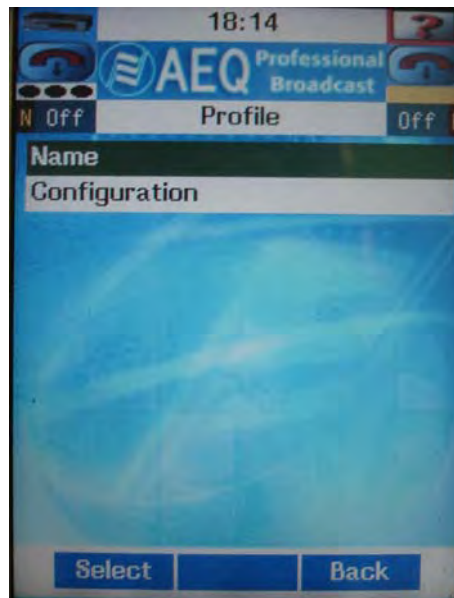
In the “Link Profiles” menu, pressing the center navigation button (associated with “Options”, visible in the lower central area of the screen) on any of the items in the list will bring up three options:

- **Add:** Add a new profile to the list.
- **Edit:** Edit a profile on the list.
- **Delete:** Eliminate a profile from the list.

3.6.1. Add a new profile to the list.

Selecting **Menu >> Link Profiles >> Options >> Add** will allow you to create a new entry in the profile list, defining its name and encoding algorithms.

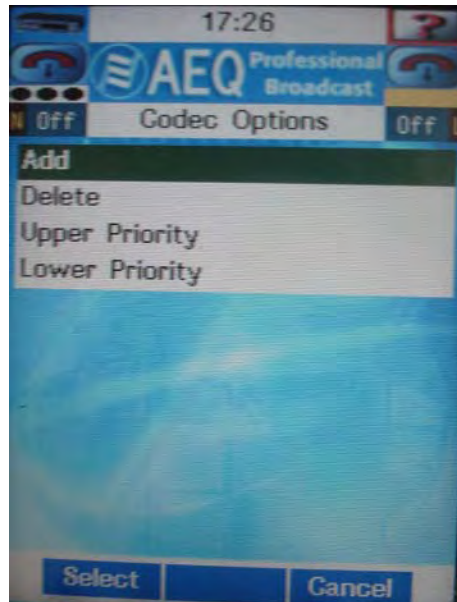
- **Name:** By using the alphanumeric keypad.
- **Configuration:** Beginning empty, you can add the encoding algorithms you wish by selecting the “Add” option repeatedly. You can enter more than one encoding algorithm in the profile and, in that case, you can assign the priority (“Upper Priority” or “Lower Priority”) and eliminate undesired modes (“Delete”).



Creating a profile

3.6.2. Editing a profile on the list.

By selecting **Menu >> Link Profiles >> Options >> Edit** you can modify the list entry on which the cursor is placed in the screen, changing its name and number, as well as the type and order of priority of the encoding algorithms it contains.



Editing the algorithms list of a profile

3.6.3. Eliminating a profile from the list.

Selecting **Menu >> Link Profiles >> Options >> Delete** will enable you to eliminate the list entry on which the cursor is placed in the screen. The system will request confirmation of this action.

3.7. Applications.

This is an advanced functions menu that includes the options for configuring the SAP option, the on-screen VU meters and the DLP option and RTP communication statistics.



Applications menu

3.7.1. SAP.

Details the list of communications logged in the SAP server. For further information on this subject, consult section 3.4.3.3 or annexe C of this manual.

From the list that is displayed in this menu, all you need to do is select an entry for which PHOENIX MOBILE will automatically tune in this transmission.

All the communications logged in this SAP server are of the multicast type, which means that the user, who selects one of them for reception, must previously have confirmed that all the network sections linking the transmitter with the receiver implement multicast capabilities. Otherwise, communication will not be possible.

The devices that receive multicast communications, according to the standard definition (model: one transmitting simultaneously to many), cannot transmit anything to the transmitter, as a one-way communication is configured.

3.7.2. VU Meters.

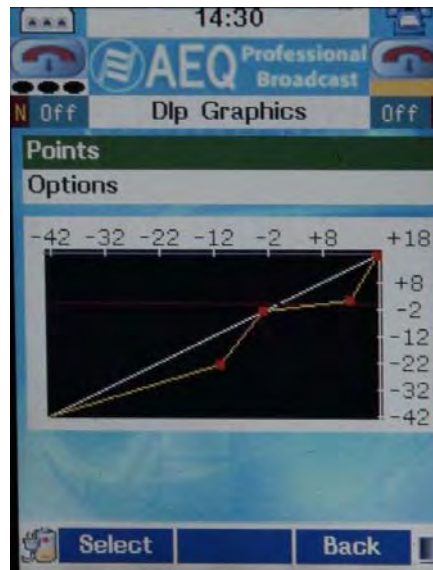
This menu allows you to select the signals you want to monitor in the VU meters on the start-up screen:

- **VU1** corresponds to the upper VU meter.
- **VU2** corresponds to the lower VU meter.

3.7.3. DLP design.

Enables you to select the desired DLP dynamics processing module operating mode from a list of available options. By default, the equipment comes from the factory with the **Linear** mode available, but you can add, edit and eliminate additional profiles.

For further information on this subject, we recommend consulting annexe D.



Graphic view of the DLP option

3.7.4. RTP statistical.

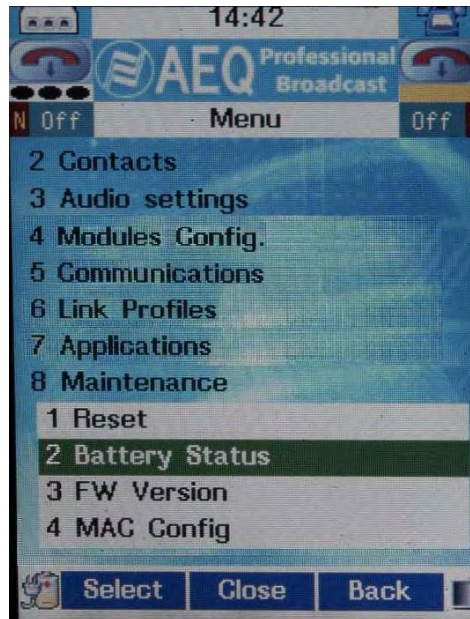
Menu devoted to the different technical parameters associated with communication over the IP interface. Communication over IP networks differs notably from the communications traditionally used to date in broadcast environments, whether they are POTS or ISDN. In IP networks there are no dedicated resources or qualities of service implemented in most systems, a situation that gives rise to variations in the delay that can occur, packet losses or the reception of disordered data (due to the internal routing inherent in IP networks based on links and router queues). This menu provides you with a reference based on values, objectively measured by the equipment, of IP communication status by means of the following parameters:

- **Lost Frame:** percentage of losses in transmitted IP packets.
- **Packet Jitter:** represents an estimate of the variation of the delay in the IP packets received, which, when these values are high, can make communication in the receiver impossible to synchronize. This information, provided in milliseconds, is important when defining the reception buffer, since the only way to reduce the jitter entails filtering it through a buffer, at the expense of lengthening the delay.
- **Delay Factor:** indicates the delay between the arrival of an IP packet and the reading of that packet from the reception buffer.
- **Media Lost Rate:** percentage of losses in transmitted IP packets that carry streaming application information.



3.8. Maintenance.

Menu of applications associated with the internal maintenance of the equipment.



Maintenance menu

3.8.1. Reset.

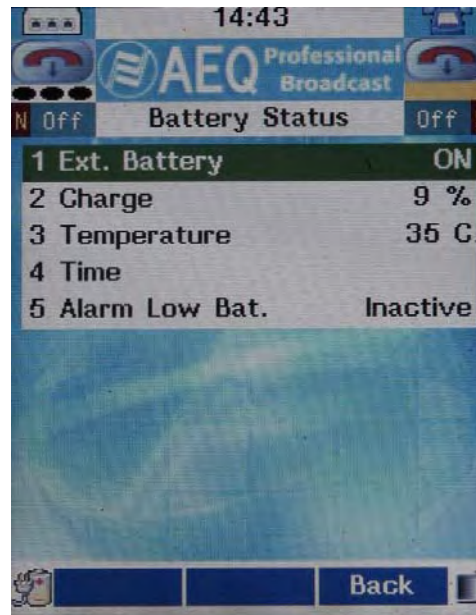
Enables to perform a physical reset of the various interfaces and integrated communication modules:

- **L Module:** Additional communication module 2.
- **R Module:** Additional communication module 1.
- **Master:** Basic communication module (IP Interface).

Selecting ON brings about the immediate reset of the interface or communications module. That field immediately shifts into the OFF status again, awaiting a future reset.

3.8.2. Battery Status.

This option is shown only when there is a battery connected to the unit. It shows information regarding the battery in the unit. When the equipment is connected/disconnected from mains power, the information contained in this menu needs a few seconds to refresh the values.

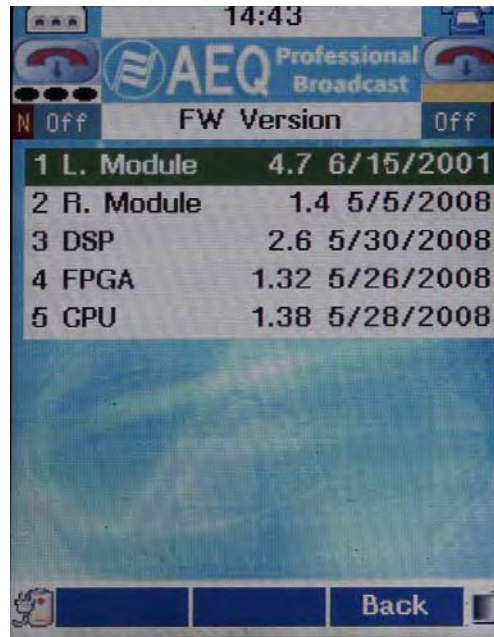


Detail of battery status

- **Ext. Battery:** ON indicates that the batteries pack is connected to mains power through the power supply/charger (this supplies power to the unit and allows the simultaneous batteries pack charging).
- **Charge:** Indicates the percentage of charge remaining in the battery.
- **Temperature:** Indicates the temperature of the battery.
- **Time:** It gives the estimate of the remaining operating time in normal mode with the remaining battery charge at that moment.
- **Alarm Low Bat.:** It indicates if the alarm associated with a low/critical available battery level is active or not. PHOENIX MOBILE will emit a warning when approximately 10 minutes of estimated battery charge remains and an alarm when one minute of available battery level remains.
In the last case, PHOENIX MOBILE blocks all the buttons, preventing any last-minute modification that, due to such a low battery level, might not be properly saved.

3.8.3. FW (Firmware) Version.

Information associated with the hardware used in manufacturing the various modules in the equipment. Essentially, this information is for AEQ's internal use; the Technical Assistance Service may ask for it in case of a breakdown or repair.

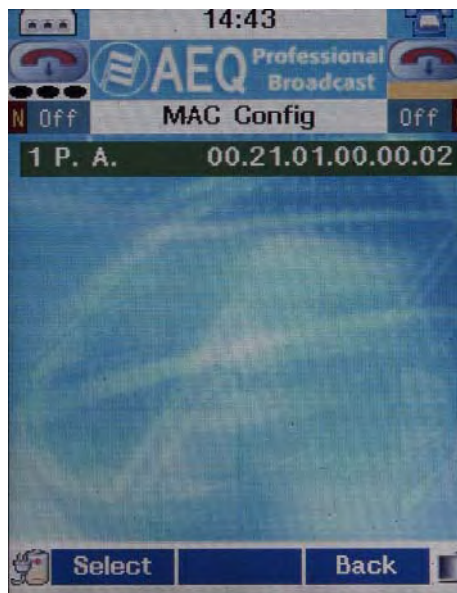


Detail of firmware menu

If a communications module is not inserted in the corresponding slot, the message “**Not Present**” will be displayed.

3.8.4. MAC Config.

Data relative to the MAC address associated with the IP interface of the codec, and which enables the address to be edited; because of the consequences this could have, the address should only be edited by highly qualified personnel or under the supervision of Authorized Technical Services, and always in possession of the required network information.



MAC Config

To edit this information, the **Select** option must be accessed by pressing the left navigator button; the information is then changed by using the alphanumeric keypad (or the center navigation button to delete).

3.9. Settings.

Menu for the internal configuration of the unit, which enables you to configure such equipment parameters as the local clock, date, screen lighting and IP configuration of the Ethernet interface of the codec.



Detail of Settings menu

3.9.1. Time and Date.

Configuration of the time and date of the equipment.



Time and Date submenu

The modification of any parameter is simply a matter of selecting the field to change with the navigator up and down arrows, pressing the **Select** option and changing the value that is displayed on the screen with the left and right navigator arrows.

3.9.2. Local Config.

Allows you to activate/deactivate the audible call notification (“**Buzzer**”) and to manage the energy saving mode associated with the screen: option 2 (“**Illumination Sleep**” or “**Power saver light**”) depending of the installed firmware versions) allows you to activate/deactivate this mode and option 3 (“**Illumination Time**” or “**Light time-out**”) allows you to set the time, in minutes, that will pass before the screen turns off when there’s no operation of the unit.



Detail of Local Config

3.9.3. Ethernet Config.

See section 3.5.1.

3.10. Users.

You can access to PHOENIX MOBILE as **Administrator** user, with access to all the configuration options, or as **Normal** user, with restricted and configurable (by Administrator) access.

PHOENIX MOBILE is supplied from the factory configured to access as Administrator user when you turn on the unit. This way, the following options are displayed in “Users” menu:

- **Logout:** allows you to leave Administrator user and access as Normal user.
- **User priorities:** allows you to select which configuration option will be available (ON) or not (OFF) for **Normal** user.
- **Password Admin:** allows you to select the password to access to the unit as Administrator user. The password is a case sensitive alphanumeric combination of up to 10 digits. The unit is supplied from the factory with “**aeqphoenix**” as default password.

When the unit is used as Normal user, the only option that appears in “Users” menu is “**Admin Logon**”, that allows you to change to Administrator user after introducing the corresponding password.

The configuration options in “**User priorities**” submenu are the following ones:

- **Contact Edit:** allows you to give access to the options of the contacts list.
- **Audio Routing:** allows you to give access to the options of the “Routing” menu (both “Crosspoint” and “Mixer” area).
- **Channel Config.:** allows you to give access to the configuration of “Program” and “Coordination” channels (relating to assignment of the interface associated to each channel).
- **3G Config.:** allows you to give access to the configuration options of “3G Config” menu.
- **SIP Config.:** allows you to give access to the configuration options of “SIP Config” menu.
- **Module Config.:** allows you to give access to the configuration options of “ISDN Config”, “POTS Config” and “GSM Config” menus.
- **Module Params.:** allows you to give access to the configuration options of “ISDN Parameters” menu.
- **Application VU:** allows you to give access to VU meters configuration.
- **IP Config.:** allows you to give access to the configuration options of “Ethernet Config” menu.

4. QUICKSTART USER'S GUIDE.

To ensure a full knowledge how the codec is used, you should first have carefully read the previous chapters. The paragraphs below describe the basic actions you will need to take to operate the equipment. If you need more detail, review the information given in the previous sections of this manual.

4.1. Connecting the equipment.

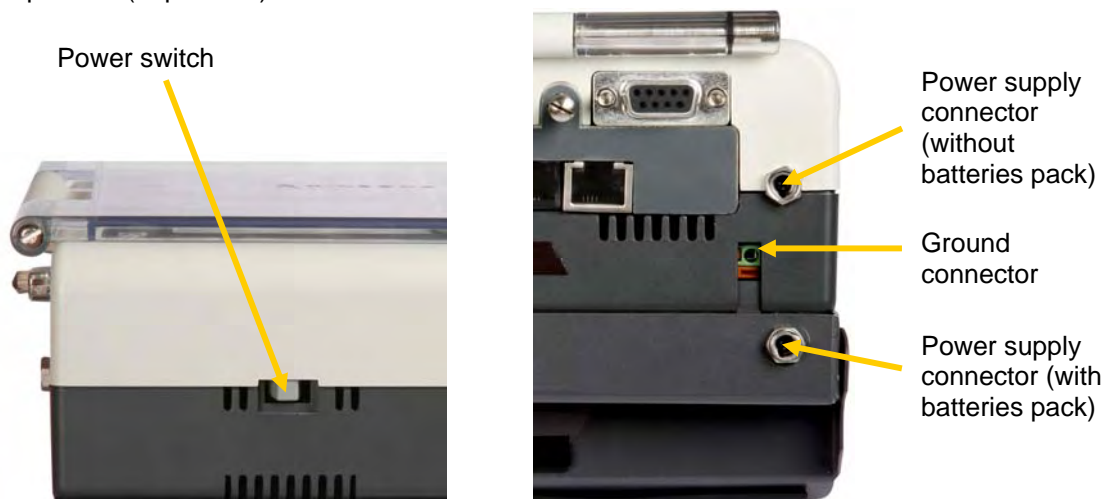
4.1.1. Power Supply.

The power supply connector (mini-XLR3p female) is located on the back panel of the codec (see section 2.2.6.1). **IMPORTANT NOTE:** In case of a batteries pack is connected to the unit, this connector is no longer operative and the one that's included in the batteries pack itself must be used.

The power switch, located on the left side of the codec, has two different positions:

- **Unpressed:** Power off.
- **Pressed:** Normal operation, with simultaneous battery charging (when battery is connected to the unit and, at the same time, to mains power through the power supply/charger supplied).

Before connecting the equipment and the charger, you must ensure that the power switch is in OFF position (unpressed).



Once the equipment and the charger have been connected, you can safely turn on the unit.

4.1.2. Audio.

Section 2 of this manual gives a detailed description of the physical connection components of the equipment. A simple approach to the connection procedure follows:

- Using XLR-3 male connectors, connect the microphones you need to the MIC1 MIC2 inputs. If you need another microphone, use the MIC/LIN input. These channels can be configured in **Menu >> Audio settings >> Input >> Mic 1, Mic 2 and Mic/Line.**
- Connect the headphones you need to the HP1 and HP2 outputs using cable terminated in a 1/4" jack. These channels can be configured in **Menu >> Audio settings >> Output >> HP1 and HP2.**
- If necessary, connect the auxiliary input/output using XLR 5-p male connector. The setup of these Aux I/O can be done in **Menu >> Audio settings >> Input >> Aux In** and **Menu >> Audio settings >> Output >> Aux Out.**

4.1.3. Communications interface.

Depending on the type of communications, a certain type of connection will be needed.

- **IP connection (Ethernet interface):** Connect an Ethernet cable (CAT5 or higher) terminated in an RJ45 10/100BT connector to the connector provided for this purpose on the back panel of the unit (see section 2.2). The cable selected will be of the flat type if the connection is made from the unit to a communications device of the switch or router type.
For further information on the pinout, see section 2.2.4 in this manual.
- **POTS connection:** Connect a standard cable terminated in an RJ11 connector to the connector provided for this purpose on the back panel of the POTS module (see section 2.5.1).
- **ISDN connection:** Connect a standard cable terminated in an RJ11 connector or an Ethernet cable terminated in an RJ45 connector to one of the two connectors provided for this purpose on the back panel of the ISDN module (see section 2.5.2).
- **GSM or 3G connection:** Connect a mobile phone to USB B port of PHOENIX MOBILE by means of mobile phone data cable and an USB B to USB A adapter.

4.2. Powering up the codec.

Once the codec has been connected to the mains, you will need to:

- Press the power switch.
- The TFT screen will light up, indicating that the equipment is receiving power from an external supply.
- If there is a battery connected to the unit and, at the same time, to mains power through the power supply/charger supplied, the “BATT CHARGING” LED, located on the control surface, will blink to indicate that the codec battery is being charged. There is also an icon in the lower right-hand area of the display indicating the battery charging process.



Detail of the battery charging indicator LED

WARNING:

1. Never tamper with the battery.
2. Always use original AEQ power supplies and, in case of failure, send the malfunctioning part to AEQ or one of its authorized agents for repair or replacement.

4.3. Review of the configuration.

Once the unit is powered up, the TFT display will show the initial or idle screen.

- Check that in the upper part of the screen the icons of the communication modules associated to Program and Coordination show **blue background** (module status is right). Regarding Ethernet interface, when the use of a Proxy SIP server as an intermediary in the communication is activated and the equipment has not been registered in the server correctly, the message “Error Register” appears.
- Go to **Menu >> Audio settings >> Routing >> Crosspoint**.
- Check that the desired crosspoints are established. If they are not, activate them. By default, PHOENIX MOBILE is supplied from the factory with the following configuration:
 - MIC1 -> TxProgram.
 - MIC2 -> TxCoord.
 - RxProgram -> HP1 (in mono mode).
 - RxCoordination -> HP2 (in mono mode).
- Go to **Menu >> Audio settings >> Routing >> Mixer**.
- Check that the sends from the crosspoints have the desired level. If they do not, establish that level.
The PHOENIX MOBILE units leave the factory configured with all the mixing factors at 100%.
- Go to **Menu >> Modules Config**. and check the communication modules associated with Program and Coordination. Modify if necessary.
By default, PHOENIX MOBILE is supplied from the factory with Program associated with the Ethernet interface and Coord. associated with the right slot.
- Verify the encoding profile to be used in communication by pressing the “**CHN**” button on the control surface. Modify if necessary.
We recommend using, for IP communication, encoding profiles that include at least one of the algorithms defined as obligatory by EBU-TECH standard 3326, to provide a 100% guarantee of success in communication (section 3.5 in this manual).
In the rest of the communication interfaces, we recommend ensuring that the encoding mode selected is compatible with the communication line and receiver unit.
- Verify the activation and configuration of the different input and output channels in the unit:
 - By manipulating the ON/OFF buttons (light on/off, respectively) on the rotary encoders for HP1, MIC1, MIC/LINE, MIC2 and HP2.
 - By using the TX COORD and TX PROG buttons on the control surface, for sending Program and Coordination.
 - By manipulating the AUX IN button on the control surface, for the auxiliary input.
 - By means of the loudspeaker icon in the case of the auxiliary output.
 - By going to **Menu >> Audio settings >> Input >> Rx Program** or **Rx Coord** for the Program and Coordination channel returns.

The status of all these selections can be directly modified through the corresponding buttons on the control surface, or in **Menu >> Audio settings >> Input** or **Output**, if there is no associated button. By default, PHOENIX MOBILE comes from the factory with the nominal gain adjustment Offset at 0dB and the input and output Gains at the nominal level of 0dB.
- Check for correct input and output audio levels through the indicators in the form of faders in the idle screen. For outgoing audio, test the microphones and use the VU meters and/or headphones to check the level. If the audio levels are not correct, modify them by using the rotary encoders on the control surface.
- Check that there is no undesired tones active in **Menu >> Audio settings >> Tones** (general reset in **Reset Tones**).

4.4. Establishment of an IP communication (Ethernet interface).

- Make sure the codec is powered up.
- Check that the RJ45 cable is correctly connected and set.
- Check the unit local IP: **Menu >> Settings >> Ethernet Config**. This information must be precise for a correct integration of the unit into the IP network over which the transmission will go out. For further information, consult with your Network Administrator.
- Check the MAC configuration of the unit IP interface: **Menu >> Maintenance >> MAC Config**. For further information, consult with your Network Administrator.
- Select the Program or Coordination channel to be transmitted over the Ethernet interface: **Menu >> Modules Config**.
- Check the equipment IP interface: **Menu >> Communications >> SIP Config >> Call Settings**.
- Check the SIP, SAP and STUN parameters of the equipment IP interface: **Menu >> Communications >> SIP Config**. For further information, consult section 3.5.3 in this manual.
- Check communication interface: **blue background** must be shown in the corresponding indicator in the up-left (Program) or up-right (Coord.) corner on the TFT screen.
- Select the encoding profile, destination and buffer size for the reception buffer by pressing the “**CHN**” button on the control surface.
We recommend using, for IP communication, encoding profiles that include at least one of the algorithms defined as obligatory by EBU-TECH standard 3326, to provide a 100% guarantee of success in communication.
- Press the **green button** on the control surface to launch the connection (the progress of the communication establishment process is displayed in the upper area of the TFT screen).
- Once the connection has been correctly established, the LED next to the transmission button corresponding to the used channel (Program or Coordination) will remain green-lighted.



Detail of established connection in Program channel

AEQ has a PHOENIX unit available for test at “phoenixMaster@sip.aeq.es” URI (IP address: 80.33.3.111) and a SIP server is also available and with warranty that it works according to the official standard.

4.5. Ending an IP communication (Ethernet interface).

- To end the communication, you need only to press the **red button** on the control surface.

4.6. Reception and acceptance of an IP communication (Ethernet interface).

If the Ethernet interface of the codec is suitably configured and the automatic answering mode is OFF:

- The unit will emit an audible notification signal. You can disable it in **Menu >> Settings >> Local Config >> Buzzer** (change it to **OFF**) or configure the length in **Menu >> Communications >> SIP Config >> Call Settings >> Auto Answer - Ringing Time**.
- The alphanumeric keypad backlighting will blink to warn the user only when “Buzzer” is OFF.

- The information on the calling URI will appear in the upper part of the TFT screen.
- Press the **green button** on the control surface to accept the call. In the upper area of the TFT screen, the synchronization status of the connection can be displayed. If you do not wish to accept the call, press the **red button** to reject it.

The automatic answering mode can be activated/deactivated in **Menu >> Communications >> SIP Config >> Call Settings >> Auto Answer - Activate.**

4.7. Establishment of a POTS communication.

- Make sure the codec is powered up.
- See that the RJ11 cable is correctly connected and set.
- Select the Program or Coordination channel to be transmitted over the POTS interface: **Menu >> Modules Config.**
- Configure the equipment POTS interface: **Menu >> Modules Config. >> [R] or [L]: Slot POTS** (depending on the slot into which this additional module has been physically inserted).
- Check communication interface: **blue background** must be shown in the corresponding indicator in the up-left (Program) or up-right (Coord.) corner on the TFT screen.
- Select the encoding mode and destination by pressing the “CHN” button on the control surface. When POTS module is assigned to **Coordination** channel, PHOENIX MOBILE allows **only Phone encoding**, simulating a telephone hybrid; when it’s assigned to **Program** channel, **AAC-LC** (12 or 22Kbps at 24KHz) and **HE-AAC** modes (12 or 22Kbps at 48KHz) can be also used.
- Press the **green button** on the control surface to launch the connection (the progress of the communication establishment process is displayed in the upper area of the TFT screen).
- Once the connection has been correctly established, the LED next to the transmission button corresponding to the used channel (Program or Coordination) will remain green-lighted.



Detail of established connection in Program channel

4.8. Ending a POTS communication.

- To end the communication, you need only to press the **red button** on the control surface.

4.9. Reception and acceptance of a POTS communication.

If the POTS interface of the codec is suitably configured and the automatic answering mode is OFF:

- The unit will emit an audible notification signal. You can disable it in **Menu >> Settings >> Local Config >> Buzzer** (change it to OFF).
- The alphanumeric keypad backlighting will blink only when “Buzzer” is OFF.
- The information on the calling number will appear in the upper part of the TFT screen.
- Press the **green button** on the control surface to accept the call. In the upper area of the TFT screen, the synchronization status of the connection can be displayed. If you do not wish to accept the call, press the **red button** to reject it.

The automatic answering mode can be activated/deactivated in **Menu >> Modules Config >> [R] or [L]: Slot POTS** (depending on the slot in which the POTS module is physically installed) **>> Off-Hook Mode.**

4.10. Establishment of an ISDN communication.

- Make sure the codec is powered up.
- See that the RJ45 or RJ11 cable is correctly connected and set.
- Select the Program or Coordination channel to be transmitted over the ISDN interface, **Menu >> Modules Config.**
- Configure the equipment ISDN interface: **Menu >> Modules Config. >> [R] or [L]: Slot ISDN** (depending on the slot into which this additional module has been physically inserted).
- Check communication interface: **blue background** must be shown in the corresponding indicator in the up-left (Program) or up-right (Coord) corner on the TFT screen.
- Select the encoding mode and destination by pressing the “**CHN**” button on the control surface.
- Press the **green button** on the control surface to launch the connection (the progress of the communication establishment process is displayed in the upper area of the TFT screen).
- Once the connection has been correctly established, the LED next to the transmission button corresponding to the channel sent (Program or Coordination) will remain green-lighted.

It's possible to change the encoding mode while the connection is established. It can be done, by pressing the “**CHN**” button and selecting the new mode in the corresponding menu; you must select the same mode in both units in order to get a correct synchronization.

IMPORTANT NOTE: When you want to establish a communication in one of the **128 Kbps** encoding modes (AEQ LD Extended or 128 MPEG LII), it's necessary to make **2 calls**: the first one is established through line 1 of the ISDN interface (first B channel) and, once it's connected, the second one is established through line 2.



Detail of established connection in Prog Channel

4.11. Ending an ISDN communication.

- To end the communication, you need only to press the **red button** on the control surface.

4.12. Reception and acceptance of an ISDN communication.

If the ISDN interface of the codec is suitably configured and the automatic answering mode is OFF:

- The unit will emit an audible notification signal. This notification can be disabled in **Menu >> Settings >> Local Config >> Buzzer** (change it to **OFF**).
- The alphanumeric keypad backlighting will blink only when “Buzzer” is OFF.
- The information on the calling number will appear in the upper part of the TFT screen.
- Press the **green button** on the control surface to accept the call. In the upper area of the TFT screen, the synchronization status of the connection can be displayed. If you do not wish to accept the call, press the **red button** to reject it.

The automatic answering mode can be activated/deactivated in **Menu >> Modules Config >> [R] or [L]: Slot ISDN** (depending on the slot in which the POTS module is physically installed) **>> ISDN Configuration >> Off-Hook L1 or L2 Mode.**

4.13. Establishment of a GSM communication (USB interface).

- Make sure the codec is powered up.
- Make sure the mobile phone is powered up and correctly connected to USB port through adapter. You should be aware that, during data transmission, mobile phone battery consumption rises considerably.
- Select the Program channel to be transmitted over the USB interface, **Menu >> Modules Config**. It's not possible to establish a **GSM** communication in Coordination.
- Check communication interface: **blue background** must be shown in the corresponding indicator in the up-left (Program) corner on the TFT screen.
- Check the USB interface configuration: **Menu >> Modules Config >> [U]: Slot GSM. PSTN Mode** must be configured as "**Voice**" ("**Data**" option will be selected only when the unit is receiving a call coming from a POTS module).
- Press the "**CHN**" button on the control surface and check that "**3G Mode**" is configured as **OFF**.
- Press the "**CHN**" button on the control surface and select destination (the number of the mobile phone connected to the PHOENIX MOBILE that will receive the call or the number the POTS module is connected to in that other unit) and buffer size for the reception buffer (values higher than 1000ms are recommended in order to avoid signal drops). In this mode, the only available encoding mode is **HE-AAC** and the bit-rate will adapt to link quality (typically 9kbps).
- Press the **green button** on the control surface to launch the connection (the progress of the communication establishment process is displayed in the upper area of the TFT screen).
- Once the connection has been correctly established, the LED next to the transmission button corresponding to the channel sent (Program) will remain green-lighted.



Detail of established connection in Prog Channel

4.14. Ending a GSM communication (USB interface).

- To end the communication, you need only to press the **red button** on the control surface.

4.15. Reception and acceptance of a GSM communication (USB interface).

If the USB interface of the codec is suitably configured and the automatic answering mode is OFF:

- The unit will emit an audible notification signal. This notification can be disabled in **Menu >> Settings >> Local Config >> Buzzer** (change it to **OFF**).
- The alphanumeric keypad backlighting will blink only when "Buzzer" is OFF.
- The information on the calling number will appear in the upper part of the TFT screen.
- Press the **green button** on the control surface to accept the call. In the upper area of the TFT screen, the synchronization status of the connection can be displayed. If you do not wish to accept the call, press the **red button** to reject it.

The automatic answering mode can be activated/deactivated in **Menu >> Modules Config >> [U]: Slot GSM >> Off-Hook Mode**.

4.16. Establishment of an IP-3G communication (USB interface).

- Make sure the codec is powered up.
- Make sure the mobile phone is powered up and correctly connected to USB port through adapter. You should be aware that, during data transmission, mobile phone battery consumption rises considerably.
- Check the 3G configuration of the unit: **Menu >> Communications >> 3G Config.** Typically, your mobile phone will be pre-configured with the operator's APN (Access Point Name) in order to function in 3G; all you need to configure in the PHOENIX MOBILE is the user name and password requested by your operator in order to have access to 3G connection and the 3G network type (UMTS or EV-DO).
- Select the Program or Coordination channel to be transmitted over the USB interface: **Menu >> Modules Config.**
- Check communication interface: **blue background** must be shown in the corresponding indicator in the up-left (Program) or up-right (Coord) corner on the TFT screen.
- Check the SIP, SAP and STUN parameters of the equipment: **Menu >> Communications >> SIP Config.** Check that **SIP Proxy** function is **activated**. For further information, consult section 3.5.3 in this manual.
- Press the **"CHN"** button on the control surface and configure **"3G Mode"** as **ON** (from this moment on you're paying for a data call). The icon shown on the TFT screen will change to the 3G icon with **yellow** background (while negotiating with your phone provider the communication parameters and IP address), turning to **red** while PHOENIX MOBILE is registering into the SIP server (this option must be active) and finally **blue** once the process is completed correctly. This process may take up to 10 seconds. When the equipment has not been registered in the SIP server correctly, the message "Error Register" appears.

IMPORTANT NOTE: In order to activate **3G** mode it's necessary that Ethernet interface is not associated to the other channel (Program or Coordination).

- Select the encoding profile, destination and buffer size for the reception buffer (values higher than 2000ms are recommended in order to avoid signal drops) by pressing the **"CHN"** button on the control surface.
We recommend using, for IP communication, encoding profiles that include at least one of the algorithms defined as obligatory by EBU-TECH standard 3326, to provide a 100% guarantee of success in communication.
- Press the **green button** on the control surface to launch the connection (the progress of the communication establishment process is displayed in the upper area of the TFT screen).
- Once the connection has been correctly established, the LED next to the transmission button corresponding to the channel sent (Program or Coordination) will remain green-lighted.



Detail of established connection in Prog Channel

AEQ has a PHOENIX unit available for test at "phoenixMaster@sip.aeq.es" URI (IP address: 80.33.3.111) and a SIP server is also available and with warranty that it works according to the official standard.

4.17. Ending an IP-3G communication (USB interface).

- To end the communication, you need only to press the **red button** on the control surface.

IMPORTANT NOTE: after that you're still paying for a 3G call; to stop that call press the "CHN" button on the control surface and configure "3G Mode" as **OFF**.

4.18. Reception and acceptance of an IP-3G communication (USB interface).

If the IP interface of the codec is suitably configured and the automatic answering mode is OFF:

- The unit will emit an audible notification signal. You can disable it in **Menu >> Settings >> Local Config >> Buzzer** (change it to **OFF**) or configure the length in **Menu >> Communications >> SIP Config >> Call Settings >> Auto Answer - Ringing Time**.
- The alphanumeric keypad backlighting will blink only when "Buzzer" is OFF.
- The information on the calling number will appear in the upper part of the TFT screen.
- Press the **green button** on the control surface to accept the call. In the upper area of the TFT screen, the synchronization status of the connection can be displayed. If you do not wish to accept the call, press the **red button** to reject it.

The automatic answering mode can be activated/deactivated in **Menu >> Communications >> SIP Config >> Call Settings >> Auto Answer - Activate**.

5. FIRMWARE UPGRADING.

PHOENIX MOBILE is supplied from the factory with the latest firmware versions available at that moment. However, it is possible than in the future, new firmware versions will appear adding new features to this equipment and you will need to upgrade the unit in order to have them available. Upgrading is a process that should be made carefully and it is recommended that this operation is done by an authorized dealer or following the instructions of AEQ's Technical Assistance Service. In case of doubt, please contact our Technical Assistance Service (sat@aeq.es).

IMPORTANT NOTE: When the **MICRO (CPU)** of the unit is upgraded, it's possible that part of its configuration will be changed: IP and MAC addresses, URI, SIP Proxy server User and associated Password. It's important to take a note of these parameters of the unit before upgrading in order to be able to configure them again after this process.

In order to upgrade the firmware of the unit, this procedure must be followed:

1. Turn off PHOENIX MOBILE.
2. Connect PHOENIX MOBILE to the PC that will be use for upgrading process by means of a crossed cable from the PC's network board to the PHOENIX MOBILE's Ethernet port.
3. Turn on the unit.
4. Open Internet Explorer in the PC and write down the IP address of the unit (<http://172.50.11.65>, for instance). The following menu will appear:



5. Go to "**FIRMWARE**" section and check if the versions that appear coincide with the official firmware.
6. If these versions don't coincide, go back to main menu, go to "**Upgrade**" section and upgrade the firmware the following way:



- a) When **MICRO (CPU)** must be upgraded:
 - 1) In "Filename to be saved..." box write down "**phxcpu.afu**". In "Filename On Your Computer" box, press "Examinar" button and load the bin file of the official version (it looks like this: "PhxCPU_Vxxx_xxxxxx_Deb.bin").
 - 2) Press "**Send File**" and check that the following message appears: "File: phxcpu.afu saved to memory". Go back to previous menu.
 - 3) Press "**Upgrade**" and check that PHOENIX MOBILE TFT screen shows the following message: "Warning. CPU upgrading!!!".
 - 4) Check that "**UPGRADE OK**" message appears in Internet Explorer and "Warning. CPU Upgraded" message appears on TFT screen.
 - 5) Turn off and turn on the unit.
 - 6) Write down the **http://172.50.11.68** IP address in Internet Explorer. Go to "FIRMWARE" section and check that MICRO has been correctly upgraded.

- b) When **DSP** must be upgraded:
 - 1) In "Filename to be saved..." box write down "**phxdsp.afu**". In "Filename On Your Computer" box, press "Examinar" button and load the bin file of the official version (it looks like this: "PhxDSP_Vxxx_xxxxxx_DAC_B.bin").
 - 2) Press "**Send File**" and check that the following message appears: "File: phxdsp.afu saved to memory". Go back to previous menu.
 - 3) Press "**Upgrade**" and check that PHOENIX MOBILE TFT screen shows the following message: "Warning. DSP upgrading!!!".
 - 4) Check that "**UPGRADE OK**" message appears in Internet Explorer and "Warning. DSP Upgraded" message appears on TFT screen.
 - 5) Turn off and turn on the unit. In Internet explorer go to "FIRMWARE" section and check that DSP has been correctly upgrade.

- c) When **FPGA** must be upgraded:
 - 1) In "Filename to be saved..." box write down "**phxfpga.afu**" (just like before). In "Filename On Your Computer" box, press "Examinar" button and load the bin file of the official version (it looks like this: "PhxFPGA_Vxxx_xxxxxx.bin").
 - 2) Press "**Send File**" and check that the following message appears: "File: phxdsp.afu saved to memory". Go back to previous menu.
 - 3) Press "**Upgrade**" and check that PHOENIX MOBILE TFT screen shows the following message: "Warning. DSP upgrading!!!".
 - 4) Check that "**UPGRADE OK**" message appears in Internet Explorer and "Warning. DSP Upgraded" message appears on TFT screen.
 - 5) Turn off and turn on the unit. In Internet explorer go to "FIRMWARE" section and check that FPGA has been correctly upgrade.

- d) When there's some **PGA-01 POTS** communications module and must be upgraded:
- 1) In "Filename to be saved..." box write down "**phxmodleft.afu**" or "**phxmodright.afu**", depending on the module is installed in the left or right slot of the unit (looking the front of it). In "Filename On Your Computer" box, press "Examinar" button and load the bin file of the oficial version (it looks like this: "PhxPOT_Vxxx_xxxxxx.bin").
 - 2) Press "**Send File**" and check that the following message appears: "File: phxpotleft.afu saved to memory" (or "File: phxpotright.afu saved to memory"). Go back to previous menu.
 - 3) Press "**Upgrade**" and check that PHOENIX MOBILE TFT screen shows the following message: "Warning. Module Left upgrading!!!" (or "Warning. Module Right upgrading!!!").
 - 4) Check that "**UPGRADE OK**" message appears in Internet Explorer and "Warning. Module Left Upgraded" (or "Warning. Module Right Upgraded") message appears on TFT screen.
 - 5) Turn off and turn on the unit. In Internet explorer go to "FIRMWARE" section and check that the module has been correctly upgrade.
- e) When there's some **PGA-03 ISDN** communications module and must be upgraded:
- 1) In "Filename to be saved..." box write down "**phxmodleft.afu**" or "**phxmodright.afu**", depending on the module is installed in the left or right slot of the unit (looking the front of it). In "Filename On Your Computer" box, press "Examinar" button and load the bin file of the oficial version (it looks like this: "PhxISDN_Vxxx_xxxxxx.bin").
 - 2) Press "**Send File**" and check that the following message appears: "File: phxpotleft.afu saved to memory" (or "File: phxpotright.afu saved to memory"). Go back to previous menu.
 - 3) Press "**Upgrade**" and check that PHOENIX MOBILE TFT screen shows the following message: "Warning. Module Left upgrading!!!" (or "Warning. Module Right upgrading!!!").
 - 4) Check that "**UPGRADE OK**" message appears in Internet Explorer and "Warning. Module Left Upgraded" (or "Warning. Module Right Upgraded") message appears on TFT screen.
 - 5) Turn off and turn on the unit. In Internet explorer go to "FIRMWARE" section and check that the module has been correctly upgrade.
- f) In Internet Explorer, go to "**FIRMWARE**" and check that all the firmware versions are the right ones now.
- g) If the **MICRO (CPU)** of the unit has been upgraded, configure the IP and MAC addresses, URI, SIP Proxy server User and associated Password the way they were initially.
- h) Turn off the unit.

6. TECHNICAL SPECIFICATIONS.*

Audio Inputs and Outputs	
Microphone Inputs	2 x XLR female. Low noise preamp. Phantom Feed. 2 K Ω .
Mic/line Input	1 x XLR female. Low noise preamp. 2 K Ω / 9 K Ω .
Aux Input / Output 4W	1 x XLR5 female. 6.6 K Ω / 50 Ω .
Headphone Output	Headphones 2 x 1/4" Jack ST with volume control
Audio Characteristics	
Clipping Level	+20dBu (auxiliary output)
Maximum MIC gain	65dB programmable, in 1dB steps
Maximum MIC level	-25 dBu
Nominal MIC level	-60 dBu
Maximum LINE/AUX gain	45 dB
Maximum LINE/AUX level	+22 dBu
Nominal LINE/AUX level	+0 dBu
Crosstalk @ 1 kHz	< -70 dB
Crosstalk @ 20 kHz	< -53 dB
Frequency response	20Hz to 20kHz +/- 0.35 dB
Total Harmonic Distortion	<0.17 % @ input -45 dBu
Equalized Noise in the microphone input @ 200 ohms	< -126 dBu
Sampling rate	
Analog I/O A/D D/A converter	24 bit Sigma-Delta 48 kHz max.
IP sampling rate	8 kHz, 16 kHz, 24 kHz, 32 kHz, 48 kHz
POTS sampling rate	8kHz without compression algorithm
ISDN sampling rate	16 kHz, 24 kHz, 32 kHz, 48 kHz
Communication interfaces	
Standard IP interface	RJ45 Ethernet Port
PGA-01 POTS Interface	POTS Modem module and telephone hybrid with frequency extender. RJ11 connector.
PGA-03 ISDN Interface	ISDN module with National-1 and EuroISDN standard.
Satellite	An external satellite telephone can be connected to the IP interface.
3G telephone	An external 3G telephone can be connected to the USB interface.
Compression Algorithm	
IP	G.711, G.722, Mpeg Layer 2, AEQ LD EXTEND and AAC.
PGA-01 POTS	AEQ POTS and AAC.
PGA-03 ISDN	G.711, G.722, Mpeg Layer 2 y AAC.
Data and control interface	
USB	Configurable USB 1.1: slave or master.
10 Base T LAN	RJ45 connector
GPI: General Purpose Input	2.7-50 V optocoupler
GPO: General Purpose Output	Open collector, Max. 50V 50 mA.

General characteristics	
Control surface	33-keys keypad 3.5" Color TFT display, 240x320 pixels
Visual indicators	2 VU meters in the display screen
Internal routing	Internal digital routing
Dimensions (without batteries)	240 x 210 x 72 mm (8.45 x 7.4 x 2.5")
Weight (without batteries)	1.65 kg (3.34 lbs)
Operating temperature range	-10 to +45 °C (14 to 114 °F)
Power supply	12V DC. Male 4-pin mini-XLR connector. Consumption 10.4VA.
Batteries pack duration	More than 2.5 hours in normal operation

*Characteristics are subject to change without prior notice.

Protection and safety standards

Marked CE.

Electromagnetic compatibility in compliance with the European Union directive: EN 50081-1, EN 50052-2.

7. A.E.Q. WARRANTY.

AEQ warrants that this product was designed and manufactured according to a Quality Assurance system that is type approved and certified in accordance with ISO standard 9001/2002. AEQ thus warrants that the required testing protocols were followed and executed to ensure the correct functionality of the product, as well as its specified technical characteristics.

Both the general design and manufacturing protocols and the particular protocols applicable to this unit are properly documented.

1.- This warranty does not exclude nor limit any of the customer's legally recognized rights.

2.- The warranty period is twelve months counted from the date on which the first customer purchases the product.

To execute this warranty, the customer must inform the authorized AEQ distributor or, in the absence of such a distributor, an AEQ sales office or the AEQ Technical Assistance Service, within thirty days following the appearance of a defect in a product that is under warranty, and provide a copy of the purchase invoice and the serial number of the product.

The AEQ Technical Assistance must, moreover, give its prior express approval of the shipment to AEQ of products to be repaired or replaced in application of this warranty.

No product returns that do not comply with these terms will be accepted.

3.- AEQ will repair a factory defective product, to include the labor required to perform the repair work, provided that the failure is caused by defects in the materials, design or labor. The repair shall be done in AEQ's Technical Assistance Service shops. This warranty does not include transport or shipment to the shop, nor return shipment.

4.- No extension of the warranty period shall be applied to products that are repaired or replaced as part of its execution.

5.- This warranty shall not be applicable in the following situations:

use that contravenes the instructions given in the user's manual; violent manipulation; exposure to damp or to extreme heat or atmospheric conditions, or sudden changes in such conditions; lightning; rust; unauthorized modifications or connections; unauthorized openings or repairs of the product; misuse; spillage of liquids or chemical products.

6.- AEQ shall under no circumstance and in no case be responsible for any type of damages, whether direct or indirect, derived from the use or the impossibility of using the product.

A.E.Q. will not be liable for the loss of information on disks that have been altered or are inaccurate, nor for any accidental damage attributable to the user or to persons handling the product.

ANNEXE A: Characteristics of the encoding modes.

G.711:

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

Bandwidth: 3.5KHz

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 16KHz, 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: 15KHz.

For further information on this subject, consult:

<http://www.itu.int/rec/T-REC-G0.722/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 64kbps encoding modes with sampling rates of 48, 32 or 24KHz, and 128kbps encoding modes with sampling rates of 32 and 48KHz.

Bandwidth: 10KHz to 15KHz.

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

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

This is the simplest existing wave form encoding mode, based on equispaced signal sampling followed by quantification of the samples in binary form.

For further information on this subject, consult:

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

AEQ LD-2:

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.

AEQ POTS:

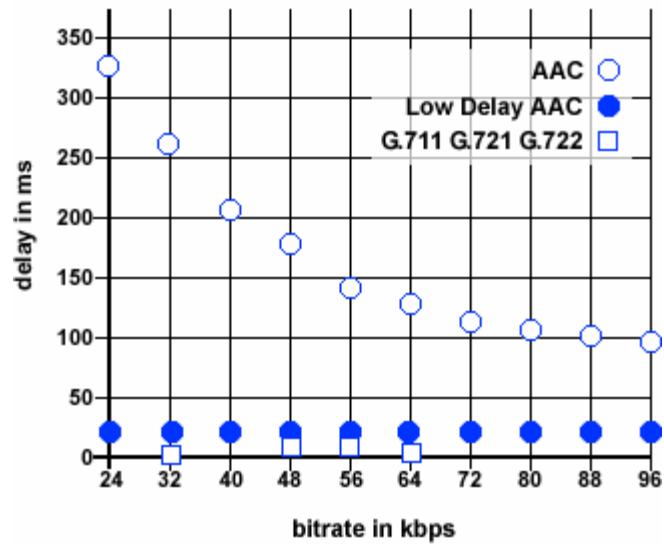
Provides standard voice calls or data transmission over analog lines.

AAC-LD:

The MPEG-4 Low Delay Audio Codec was designed to combine the benefits of perceptual audio encoding with the low delay needed in two-way communications. It is a variant of the MPEG-2 Advance Audio Coding (AAC) format.

Bandwidth: 20KHz

For further information on this subject, consult: ISO/IEC 13818-7:1997 and ISO/IEC 13818-3.

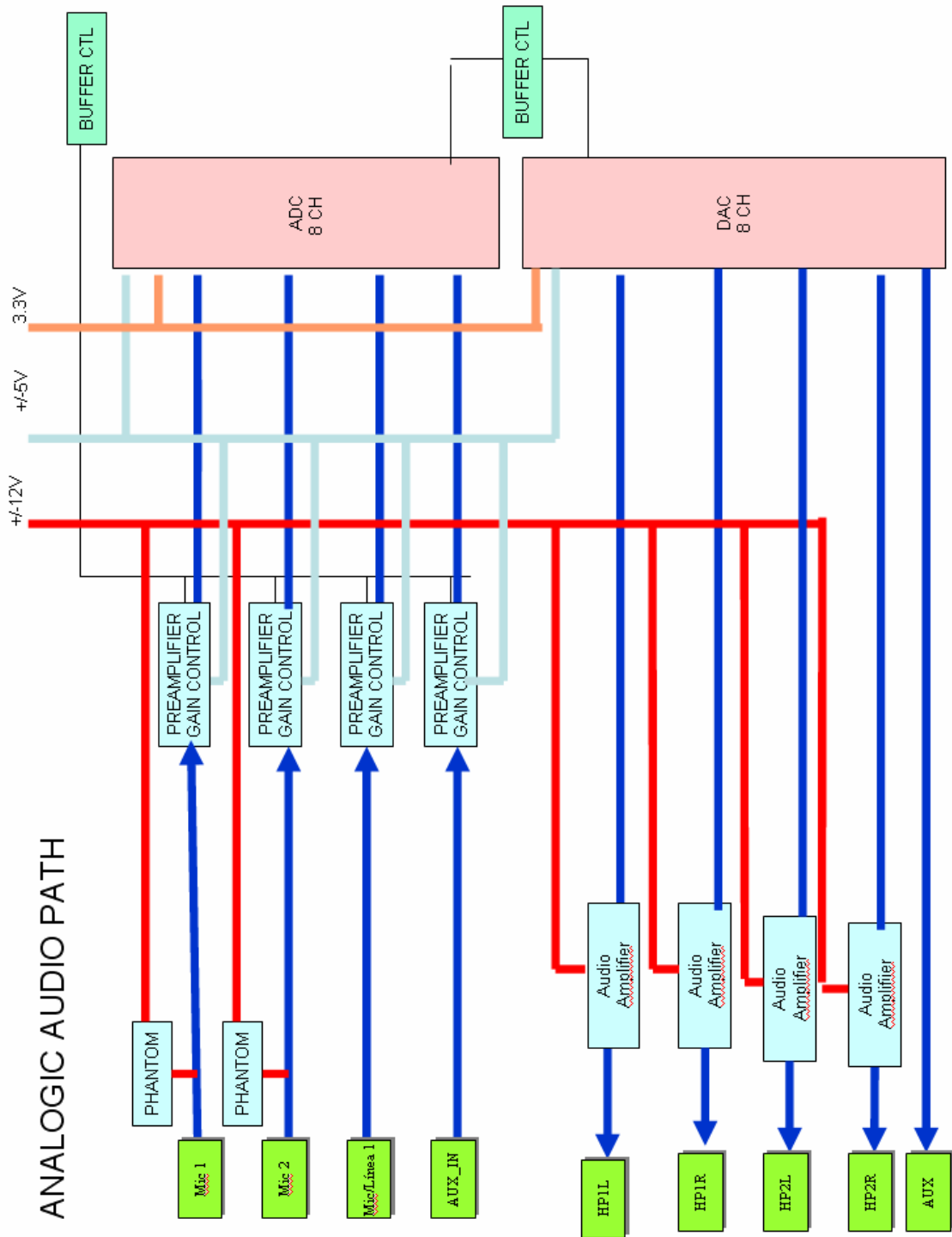


Comparative graph of different algorithms

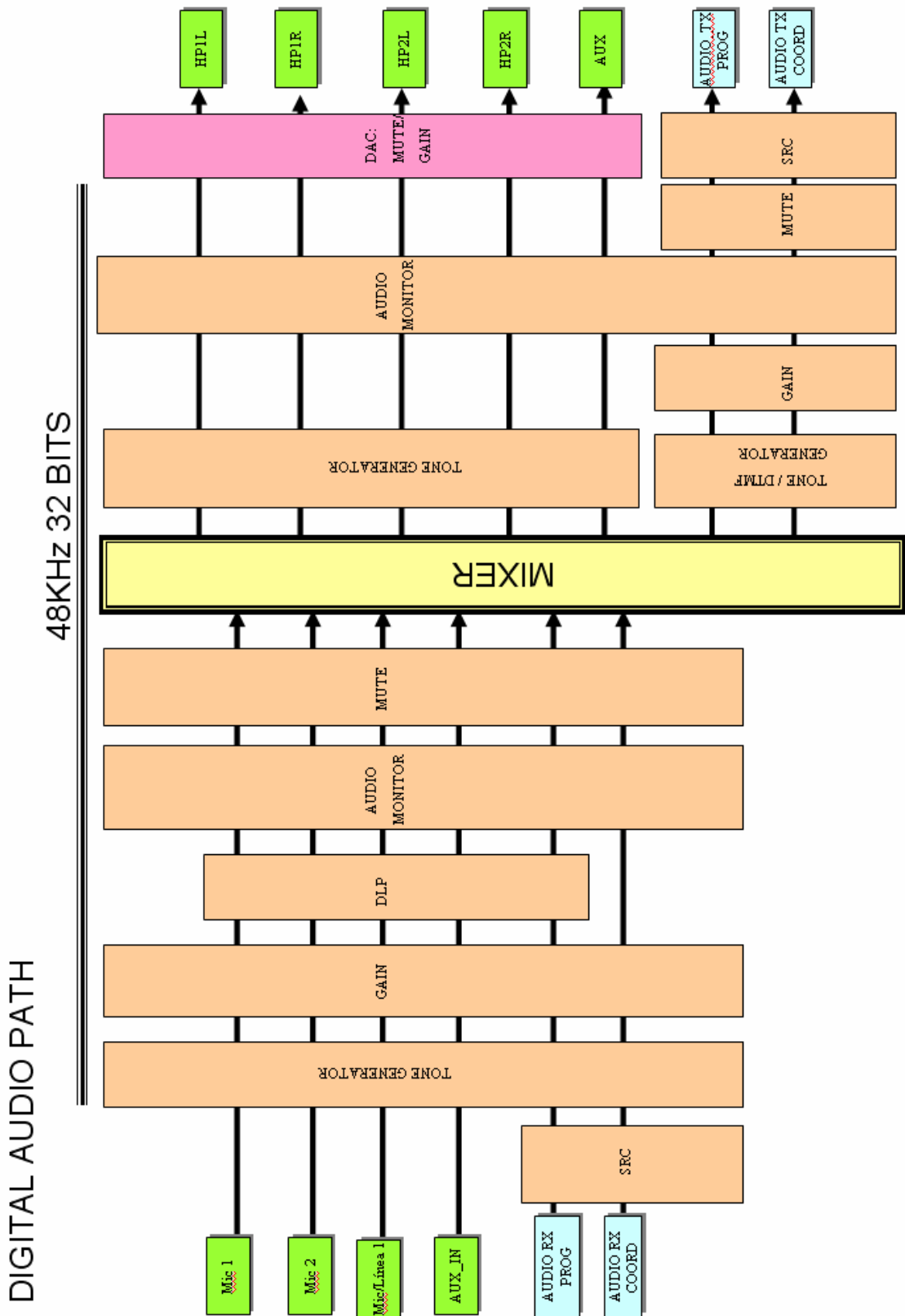
AAC-HE v1 (AACplus):

High Efficiency AAC is a compression method with optimized loss for digital radio applications. It is an extension of Low Complexity AAC (AAC-LC) optimized for low bit-rate applications such as audio streaming. Version 1 was standardized in 2003 by the MPEG group and was published under the aegis of MPEG-4 in the document ISO/IEC 14496-3, Amd.1:2003.

ANNEXE B: Detailed block diagrams.



Detailed block diagram of the analog audio part



Detailed block diagram of the digital audio part

ANNEXE C: Protocols associated with communication over IP networks.

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 signaling, establishment, maintenance and end.

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

- Signaling, 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 (RTP) and 5005 (RTCP).

While this annexe 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 MOBILE user and, as a result, the operation of this equipment. The user interested in expanding his or her knowledge of some or all of these subjects is encouraged to turn to the extensive, excellent technical material currently available regarding the IP realm and the technologies associated with it.

C1. Circuit switching vs. packet switching.

The communications systems traditionally used in the broadcast environment for applications with portable codecs have been mostly telephone or ISDN networks that are circuit switching networks; PHOENIX MOBILE, on the other hand, uses a packet switching network.

C1.1. 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 POTS.

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

C1.1.1. Benefits:

- 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 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.

C1.1.2. 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.

C1.2. 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.

C1.2.1. 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.

C1.2.2. Benefits:

- 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.

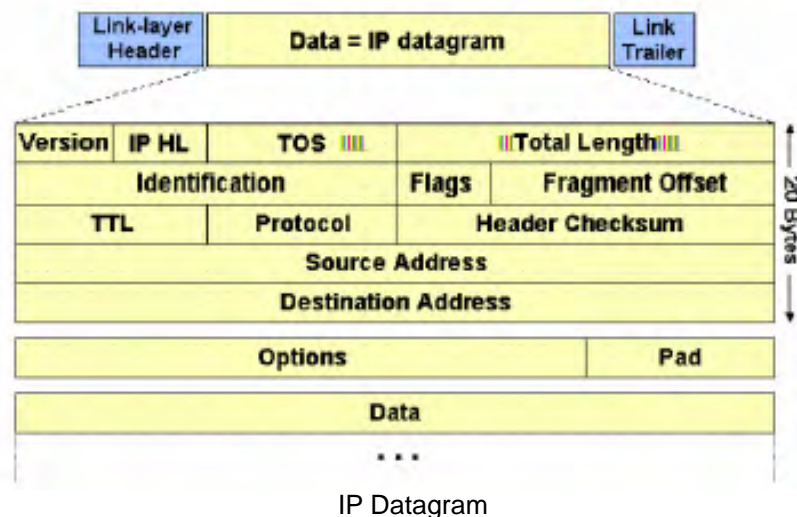
C1.2.3. Disadvantages:

- 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.

C2. IP protocol.

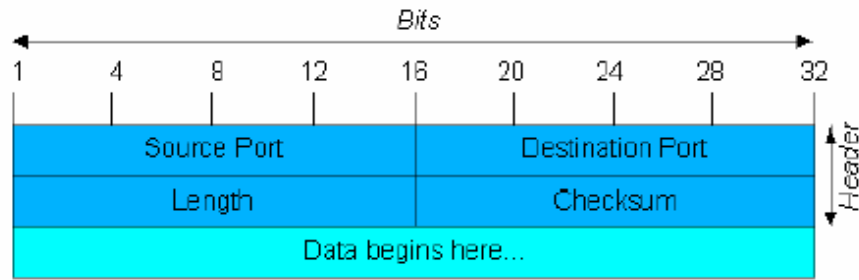
The Internet Protocol (IP) is a non-connection oriented protocol used both by the origin and the destination in data transmission over a switched packet network.

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).

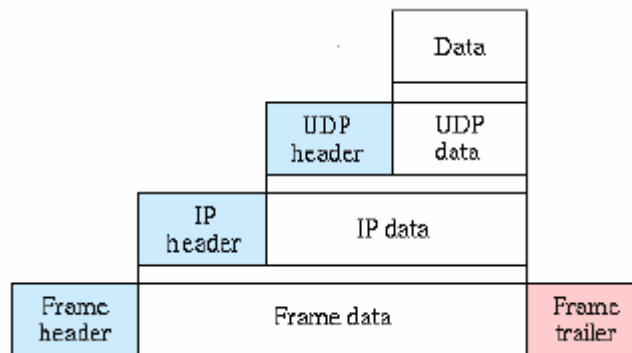
Reliability over TCP is obtained through the use of retransmissions. Real-time applications such as an audio link, with the timing requirements inherent in the information contained in the link, do not offer any useful guarantee. Since the data that are not received, and whose retransmission is requested 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

The protocols for transport over IP, 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, AEQ recommends consulting:
<http://tools.ietf.org/html/rfc791>
<http://www.iana.org/assignments/port-numbers>



IP Encapsulation

C2.1. IP addresses.

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.

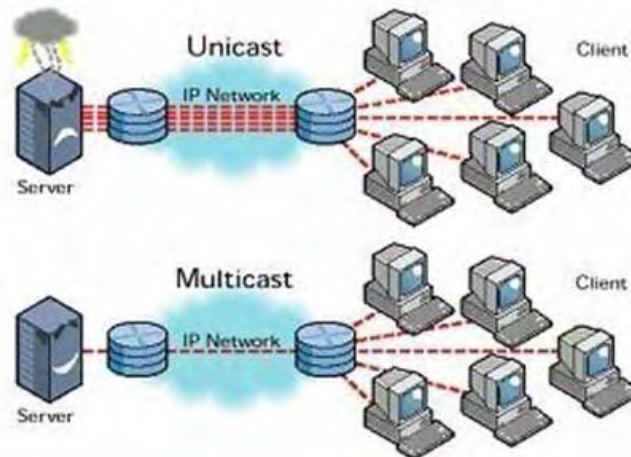
Another possible IP address classification can be made according to address validity:

- 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.

C2.2. 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.



Graphic 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.

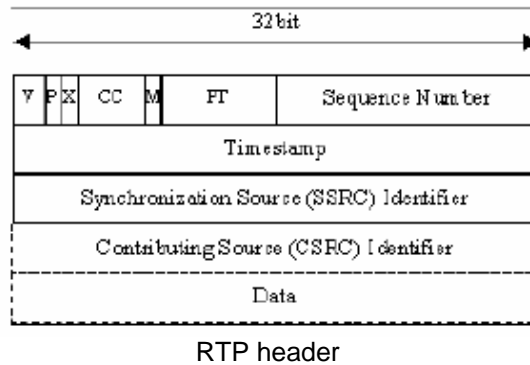
C3. RTP protocol.

RTP is the abbreviation for 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).

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.



For complete information on RTP/RTCP protocol, AEQ recommends 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>

C3.1. PHOENIX MOBILE default configuration.

PHOENIX MOBILE is an IP audio coder that operates by using RTP over UDP in IP version 4.

By default, PHOENIX MOBILE is supplied from the factory with the following IP ports defined: 5004 for RTP and 5005 for RTCP if this protocol is used.

C4. 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 where multimedia elements such as video, voice and instant messaging are involved.

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 without aftercost the services of proprietary SIP server, though 100 % of the time can neither guarantee functioning nor make responsible for the disadvantages that could suppose therefore the final user. The unit goes out of factory preconfigured with the necessary parameters to work using the resources of this SIP server (see "Default SIP configuration for PHOENIX MOBILE" in this manual).

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.

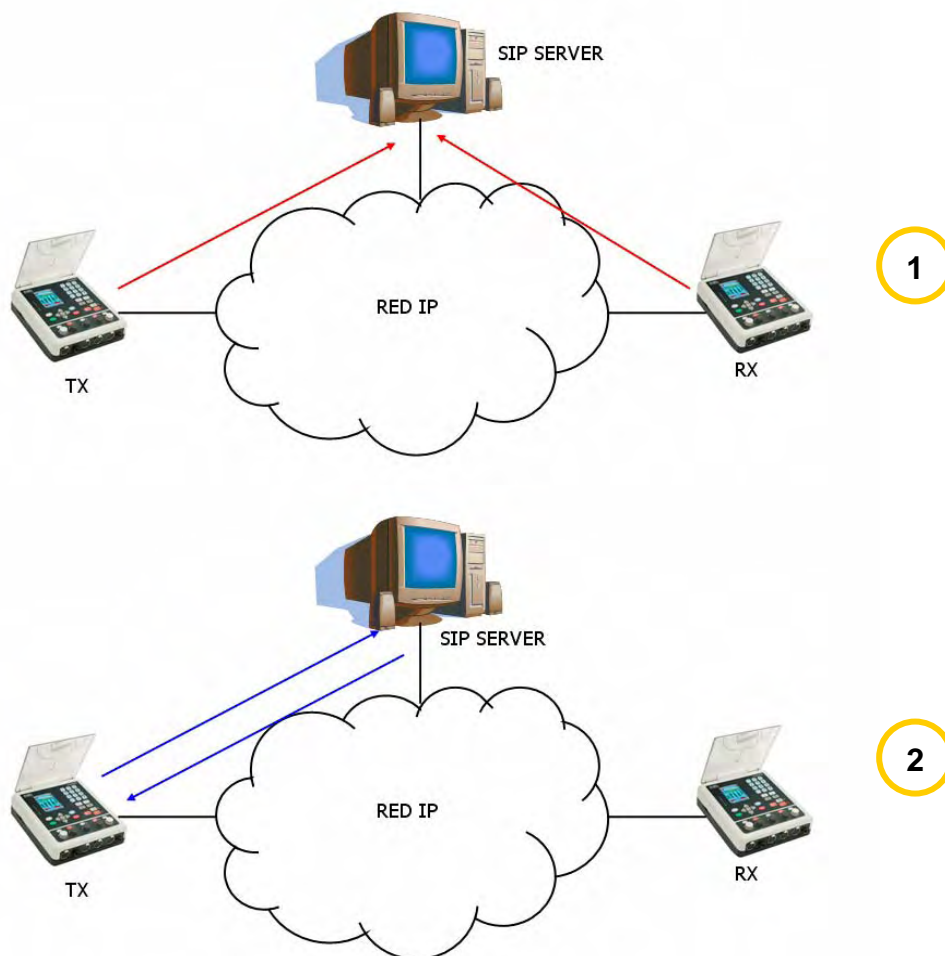
SIP also contributes register functions that enable the user to inform the proxy servers of his or her location.

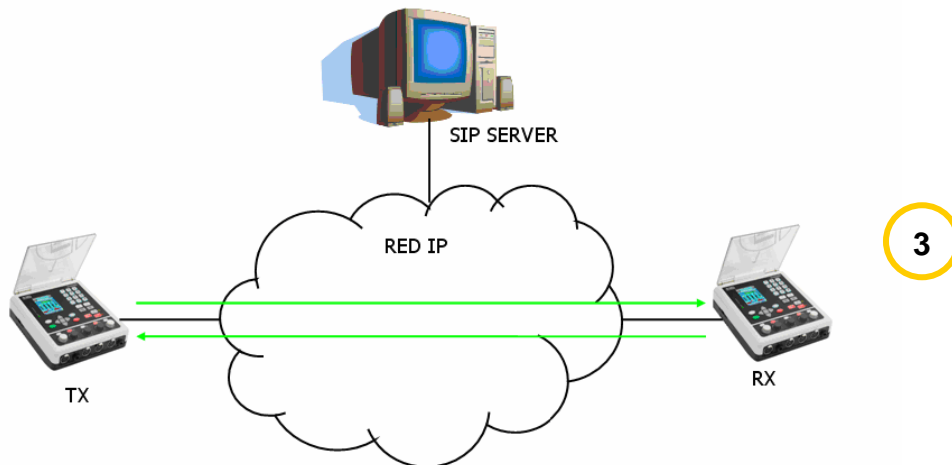
For complete information on the SIP protocol, AEQ recommends consulting:
<http://tools.ietf.org/html/rfc3261>

C4.1. Working modes.

With the SIP option activated in the PHOENIX MOBILE, 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 functional diagram. Phase 1: Register. 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 communication phase, the encoding algorithm is negotiated simultaneously, based on the Link Profiles defined in each of the devices at the two ends of the connection circuit.

C4.2. Possible work scenarios.

Depending on the type of network to which the PHOENIX MOBILE is connected, the codec will have one or another type of IP address available to it.

- Static public IP addresses offer the ideal situation, since 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 MOBILE operates perfectly with an associated SIP server and equipment identifiers of the name@domain type, and even with an associated SIP server with a URI of the name@IP-address type if the device on the opposite end of the communication circuit also has an IP address of the same type. This situation, as we have remarked earlier, is unusual due to the high cost and scant number of available public IP addresses.
- Dynamic public IP addresses allow the use of URIs of the name@domain or name@IP-address type, but it is advisable always to work with an associated SIP server, since the IP address assigned to the equipment can change each time the user powers up the unit.
- Private IP addresses, both static and dynamic, 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 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.

C4.3. PHOENIX MOBILE default configuration.

To simplify the operation of the unit, AEQ offers without aftercost the services of proprietary SIP server.

PHOENIX MOBILE is supplied from the factory with the following configuration:

- URI: of the phoenix_x@sip.aeq.es type
- SIP Proxy Activate: ON
- SIP Proxy Address: 213.97.205.219
- SIP Proxy Authenticate User: phoenix_x type
- SIP Proxy Authenticate Password: an alphanumeric combination of 8 digits
- SIP Proxy Authenticate Realm.: sip.aeq.es
- SIP Proxy Register Domain: sip.aeq.es
- SIP Proxy Register Expires: 1 minute

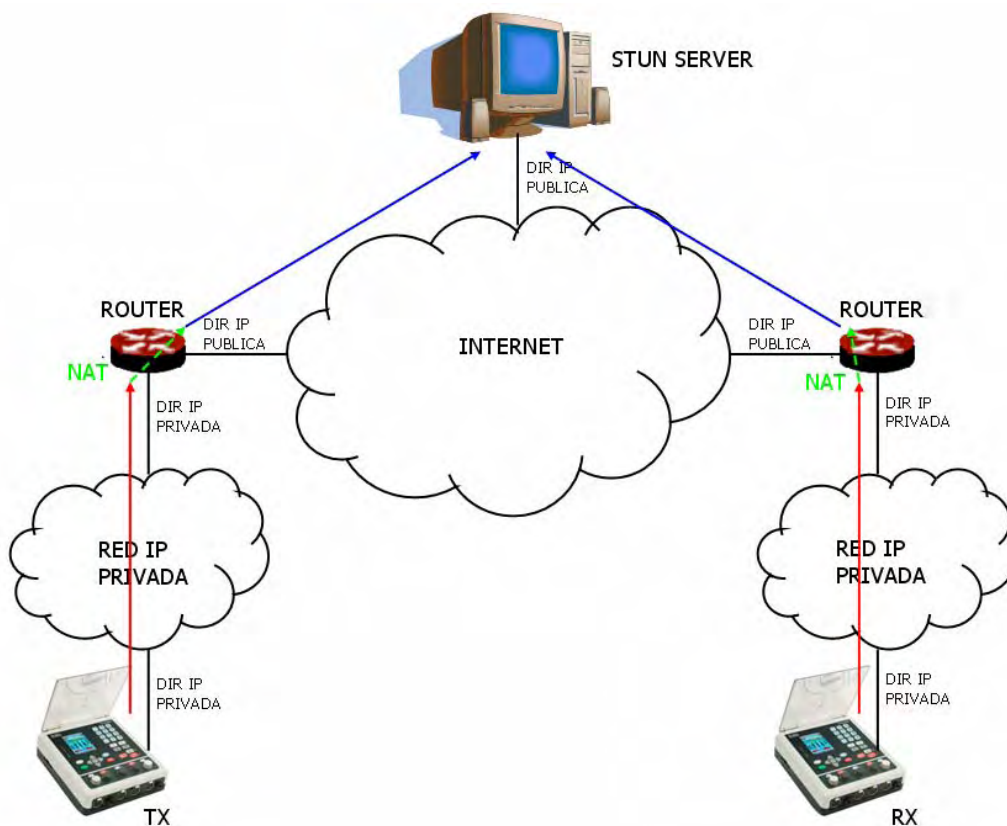
This configuration is the correct to use AEQ SIP server.

C5. STUN protocol.

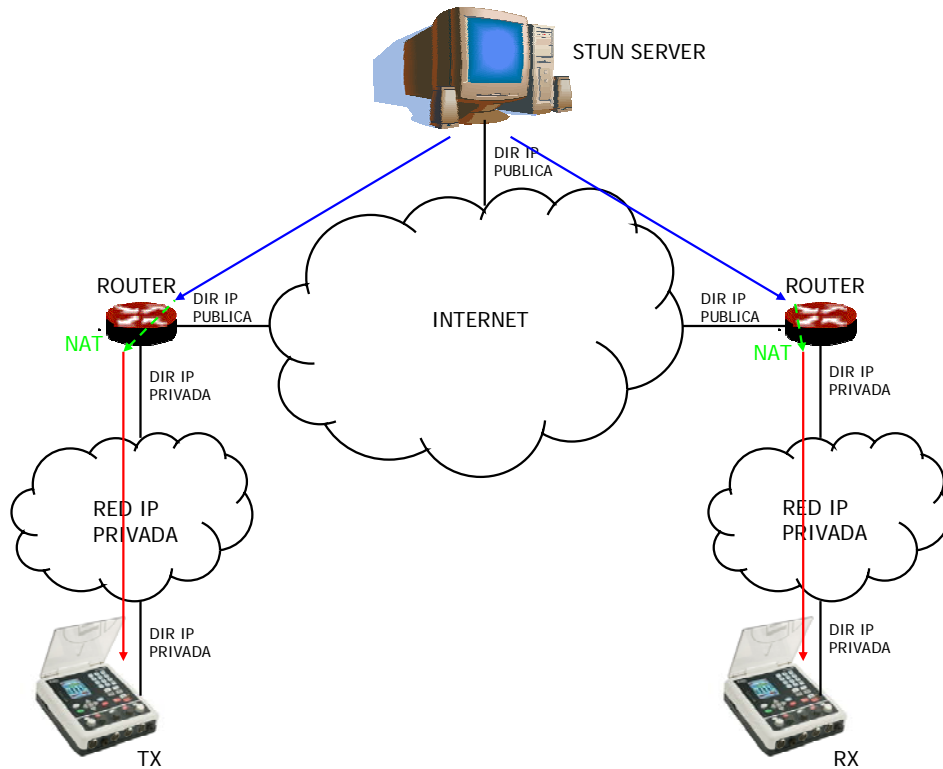
STUN (Simple Transversal of UDP over NATs) is a network protocol of the client/server type that enables NAT clients to find their public IP addresses, the type of NAT they are found in and the Internet port associated with the local port through NAT. This information is used to configure a UDP communication 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 by a gateway/router.

PHOENIX MOBILE 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 MOBILE to identify its position within the SIP server.



Register 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 PHOENIX MOBILE 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, AEQ recommends consulting:
<http://tools.ietf.org/html/rfc3489>

C5.1. PHOENIX MOBILE default configuration.

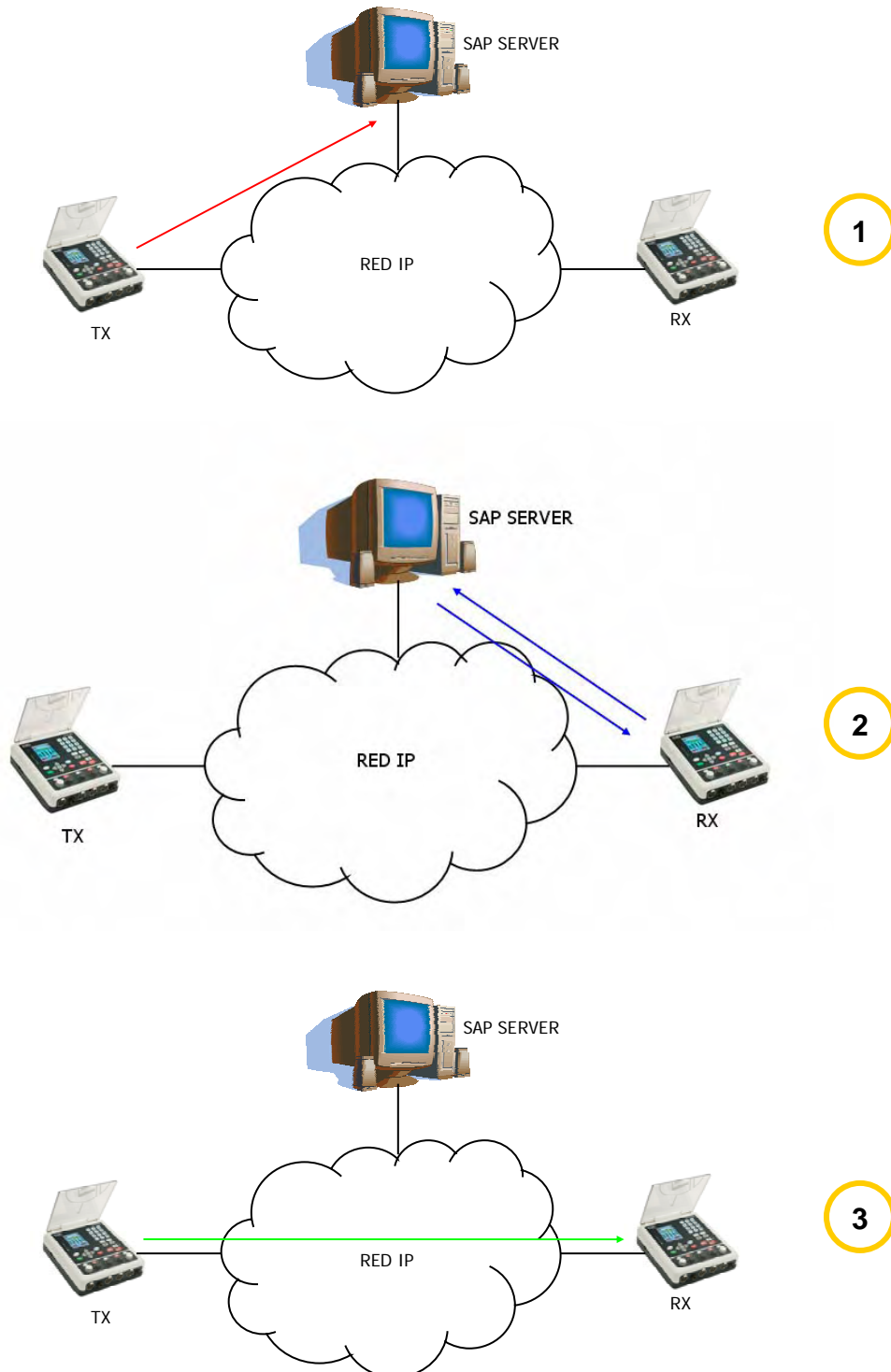
To simplify the operation of the unit, PHOENIX MOBILE offers the possibility to work with an associated external valid STUN server, using default configuration, supplied from the factory:

- Stun Protocol Activate: OFF
- Stun Protocol IP: 064.069.076.023
- Stun Protocol Port: 3478

C6. SAP protocol.

The Session Announcement Protocol is a protocol used to issue information relative to multicast sessions.

There is a SAP server that centralizes the information corresponding to the sending devices that are available and transmitting at any time, where the receiving units locate the desired transmission.



SAP protocol functional diagram. Phase 1: Transmission register. Phase 2: Query by receivers regarding active transmissions. Phase 3: Establishment of communication

SAP typically uses the Session Description Protocol (SDP) as a format for describing sessions, and multicast sessions typically use RTP as a transport protocol.

The SAP protocol is almost entirely based on multicast type transmissions, which makes it practically unviable in Internet environments, and limits its application to environments or sections of generally private networks with multicast features implemented.

For complete information on the SAP protocol, AEQ recommends consulting the document IETF-RFC 2327.

C6.1. PHOENIX MOBILE default configuration.

To simplify the operation of the unit, PHOENIX MOBILE offers the possibility to work with an associated external valid SAP server, using default configuration, supplied from the factory:

- SAP Activate: OFF
- SAP IP: 239.255.255.255

C7. Summary.

We recommend activating the SIP, SAP and STUN options because they greatly simplify the operation of the unit, so that when the PHOENIX MOBILE is powered up and is connected to an IP network, it will automatically perform the following actions:

- Search for the STUN server and register itself, thereby succeeding in discovering the available NATs.
- Look for the SIP server and register itself there with its name and position (by using the information obtained in the previous step), thus succeeding in making its position independent from its URI when connections are established.
- If a multicast transmission is to be made, it will search for the SAP server and will register there, thereby announcing its presence to the rest of the interested users.

AEQ has a PHOENIX unit available for test at “phoenixMaster@sip.aeq.es” URI (IP address: 80.33.3.111) and a SIP server is also available and with warranty that it works according to the official standard.

Protocol	Source Port	Destination : port
ICMP (ping)		64.233.183.99
SIP	5060	213.97.205.219:5060
RTP	5004	80.33.3.111:5004
RTCP	5005	80.33.3.111:5005
STUN	5004, 5005 and others	64.69.76.23:3478 and others

Table 3. IP Protocols

ANNEXE D: Digital Level Processor (DLP).

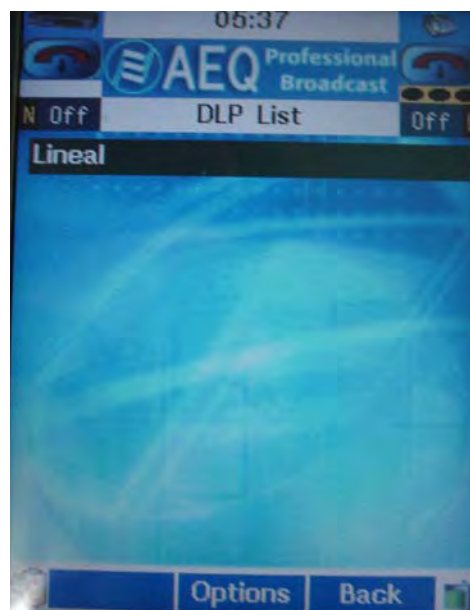
The digital level processor (DLP) is an additional feature included in the PHOENIX MOBILE that was developed to improve the quality of the processed signal by modifying its level. According to the configuration, the DLP will make it possible to:

- increase the program signal level.
- apply compression to the program signal.
- apply a limiter to the signal.
- apply an ambient noise reduction effect.

By default, the unit leaves the factory configured with a pre-programmed profile, called Linear, although the user can create, edit and eliminate new profiles in **Menu >> Applications >> DLP Design**.

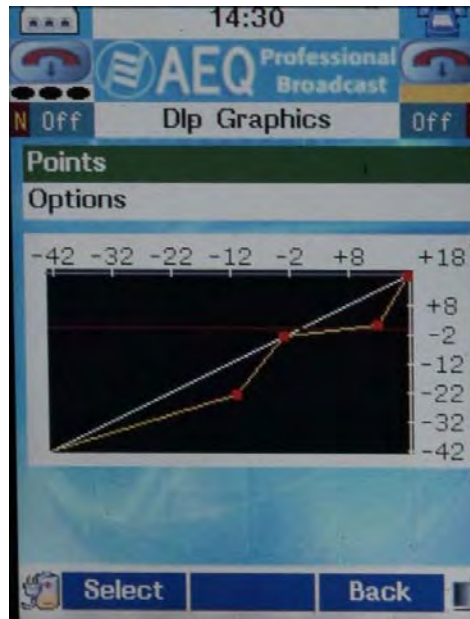
While it is not an especially complex process, the creation, configuration and modification of a DLP profile is an operation that is seldom performed. For this reason, we have included it in an appendix of the manual. Moreover, the ability to read and understand DLPs is only required by technical personnel responsible for the technical support of the equipment; these technicians, then, will be responsible for correct DLP configuration.

The activation of digital level processing is such an elementary function (easily accessible to users whether they are technically qualified or not), that it has been described previously in those points of the manual where its use is recommendable, and thus is not treated in this annexe.



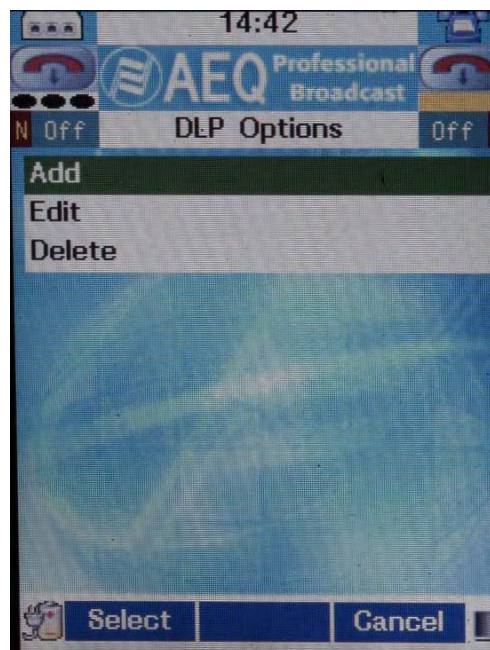
Detail of DLP Design Menu

The Linear profile supplied by default with the unit is not specifically useful, in the normal operation of the codec, inasmuch as its configuration basically results in the input signals being processed so that they retain the same levels at the output as they had on the input side (see next figure). This default profile cannot be edited, but it serves as a reference to the user for the subsequent creation of other profiles according to the specific needs of each application and client.



Graphic representation of a DLP profile

In **Menu >> Applications >> DLP Design**, the user can display the list of available profiles and make use of the option associated with the center navigator button (**Options** on the screen) of adding new profiles and modifying or deleting existing ones.



DLP Design menu options

D1. Add.

Enables you to add a new DLP profile. The fields you need to fill in are:

- **Name:** It's the name you want to give to the new profile. This is the identifier of the created profile, by means of which the activation is performed from the various menus where the DLP is accessible. Key in the name with the alphanumeric keypad.
- **Upper & Lower regions:** Enables you to configure the processor attack and release times for the two areas of action.
 - Lower region: the region that is under the transition level.
 - Upper region: the region that is above the transition level.

- **Graphics Config.:** It shows a graphic view of the DLP, with its corresponding configuration.



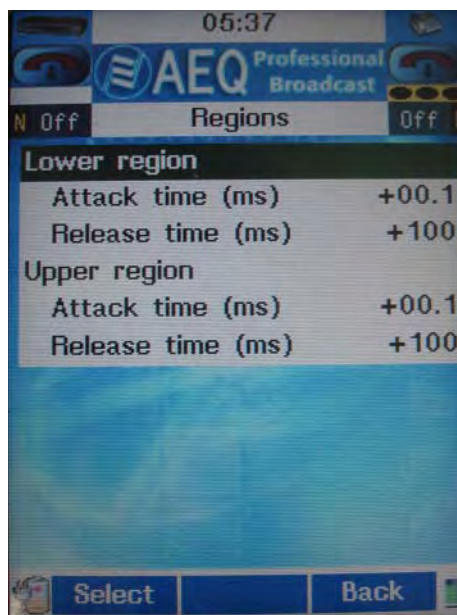
Fields to be completed in a new DLP

At the end of the process, PHOENIX MOBILE asks for your confirmation before recording the new DLP profile you have created in the internal memory. “Save” corresponds to the center navigator button.

D1.1. Upper & Lower regions.

This submenu enables you to configure the processing attack and release times for the two working regions of the equipment, separated by the transition level to be defined.

- **Attack Time:** The time that elapses from the detection of the peak to the application of the effect.
- **Release time:** Recovery time. Time elapsing after the peak during which the effect is maintained.



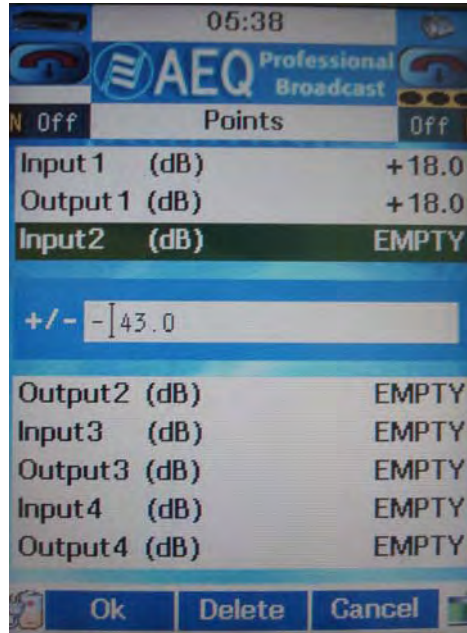
Upper & Lower regions submenu

D1.2. Graphics Config.

Enables a graphic view of the response of the DLP, and of its configuration.

D1.2.1. Points.

In this submenu you can select as many as four points that configure the response to the DLP processor. For each of the points you will have to establish the associated input and output levels.

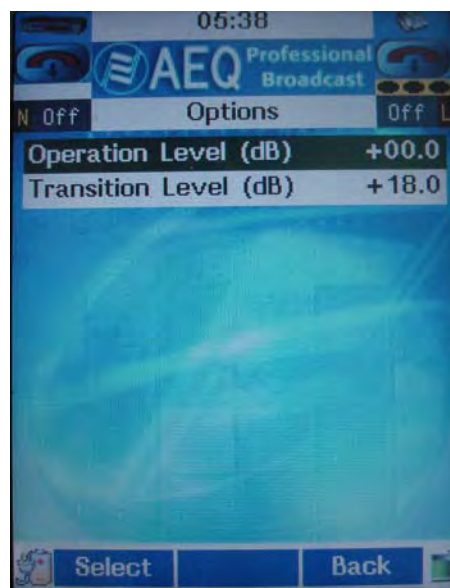


Points submenu

D1.2.2. Options.

Basically, the Options submenu lets you define the operating and transition levels. The working units are always dB.

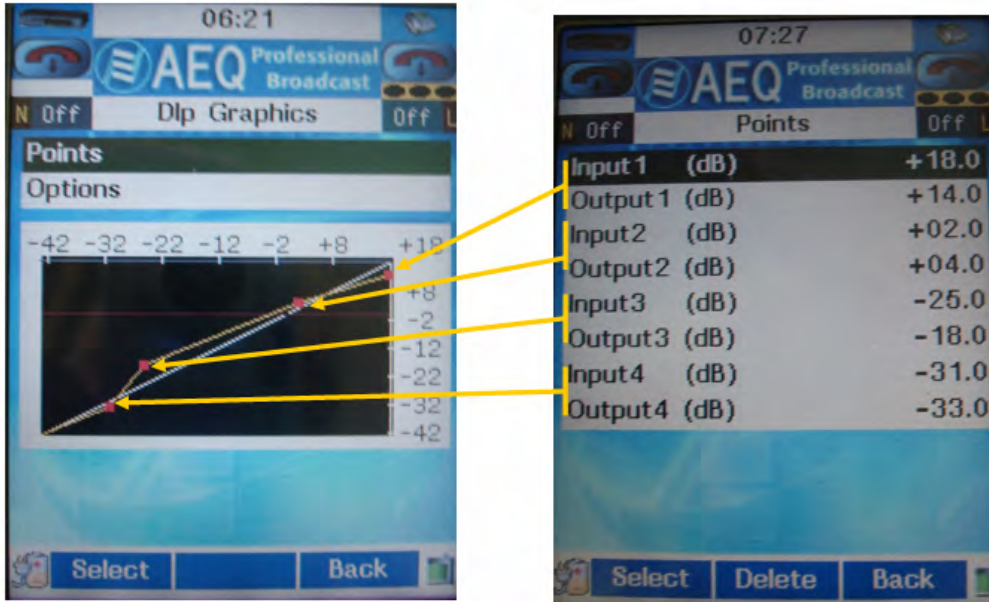
- **Operation Level:** Operation level. Defines the area of the level spectrum in which you are going to operate. Basically, this is not useful.
- **Transition Level:** Transition level. This is a key point in DLP operation. The transition level separates the point where you wish to distinguish a limiter from a compressor and / or expander.



Options submenu

D1.2.3. Example of a configuration.

The following lines describe a specific configuration as an example of the operating mechanism implemented in the unit. The values given in this example are merely orientative; it is understood that it is the user who, depending on the specific circumstances of the application, will establish the parameters needed to fulfill his or her requirements.



Graphic view of the DLP given as an example, with the associated points

In the example, the audio signal in TX that comes in with levels below -31dB will be attenuated, which means that the ambient noise is being attenuated. From -31 dB input level, the signal is strongly amplified, until it reaches the point where -25dB of input produce -18dB on the output side. In the area between -25 and $+2\text{ dB}$ input, the curve evolves from a 7dB gain to a 2dB gain (amplification). From the 2dB input level, the gain is progressively reduced (compression) so that, when the level reaches $+18\text{dB}$, the signal is reduced by 4dB . The transition level separates the compression area as the 'Upper zone' from the rest, designated 'Lower zone'; thus, the attack and release parameters edited in the previous figure would be applied to each of the two zones.

The fundamental difference between a compressor and a limiter lies in the attack and recovery times. While a limiter is conceived to eliminate those infrequent, sporadic peaks in the signal that may saturate it, a compressor is designed to "hold" a high signal modulation in constant mode. Thus, the functional principle of the expander is the same as for a compressor, but with an essential difference: the expander gains signal in constant mode for those low modulations at all times. We can thus further define a limiter by pointing out that it has short attack and recovery times in order to act quickly and effectively on those signals that may produce saturation. A compressor should have a longer attack time, and an even longer recovery time, in order to achieve a controlled gain at all times. The same applies to the expander.

D2. Edit.

From the Edit menu you can manage the modification of a previously created DLP profile. The configuration options are basically the same as when you create a new profile; we therefore recommend that you consult the previous section and review the steps to follow to make any change.

D3. Delete.

This option enables you to eliminate the currently selected DLP profile. The system will ask for your confirmation.

ANNEXE E: Connect PHOENIX MOBILE into ISDN National-1.

Practical notes for the configuration of the ISDN TERMINAL ADAPTER (TA) IN THE UNITED STATES AND OTHER COUNTRIES WITH NATIONAL 1 ISDN PROTOCOL:

The ISDN (Integrated Services Digital Network) is a telephone connection system that permits the establishment of fully digital end-to-end communications. There are two types of ISDN links, depending on the number of 64 Kbps channels: Basic links (BRI – Basic Rate Interface) and Primary links (PRI – Primary Rate Interface):

- **Basic links** allow the simultaneous establishment of two connections or data channels, each with a capacity of 64Kbps.
- **Primary links** allow 23 - 64 Kbps channels to be established in the USA, and 31 - 64Kbps channels in Europe.

To access ISDN services, you must subscribe to an ISDN phone line and connect equipment to it, such as an audio codec, that has an ISDN Terminal Adapter device to perform the tasks of establishing and maintaining connections through digital ISDN phone lines.

The PHOENIX MOBILE has an ISDN Terminal Adapter for Basic links (BRI), which gives it a communications capacity of two independent 64 Kbps data channels.

We can find two types of physical connection to the ISDN: connection to a type U interface, and connection to a type S/T interface:

- The **U interface** connects the telephone exchange with the subscriber. It has two wires, the same as in conventional analog telephony, and only allows a point-to-point connection (a single device connected to the interface).
- The **S/T interface** is obtained from the U interface, passing it through a Network Terminator (NT1) located at the subscriber's premises. It has four wires and allows a multipoint connection of up to 7 devices.

In the USA, phone companies do not supply the NT1, so the user is the one who must install it if he wants it. The most common situation is for there to be no NT1, and therefore access is through a U interface.

In summary, in the USA, access via a U interface is most common, unless the subscriber has asked the phone company to install an NT1 or has installed it himself (the NT1 is usually a box attached to the wall, with a power supply, and some kind of light indicator, to which the two-wire external line enters and which has RJ-45 jacks).

The PHOENIX's universal TA (terminal adapter) allows both types of connections, and thus has two ISDN connectors, one RJ-45 (8 pins) for the S/T interface connection, and one RJ-11 (6 pins) for the type U interface connection. The type of connection used must be configured in the unit's setup.

Although the ISDN is supposed to be a standard, differences can be found between the protocols used by different countries. In the USA, there are several protocols that are incompatible with each other; fortunately, however, there is one protocol that is fairly standardized (called National-1), which is usually available at all exchanges.

The PHOENIX's Universal Terminal Adapter can work with the NATIONAL-1 protocol (the most widely used in the USA). Selection of the desired protocol, in our case the NATIONAL-1, must be configured in the unit's setup.

In Europe and in other geographic areas, there is a protocol called EURO ISDN. There are minor differences between the EURO ISDN in different countries. These differences do not affect the PHOENIX MOBILE.

E1. Directory Numbers (DN's).

These are the telephone directory numbers assigned to the line.

In the **National-1** protocol, there will be two numbers, each associated with one of the line's channels. These numbers must be programmed in the unit for it to work correctly ("NUMBER LINE X" option) and therefore, each channel will only respond to its associated number (Note: In the USA, the directory number that must be programmed in the terminal should not contain the area code, in other words, use only the seven-digit local number).

If the directory numbers are programmed incorrectly in the unit, it will be able to make outgoing calls but will not respond to incoming calls.

E2. SPID's (Service Profile Identification Numbers).

In the case of the NATIONAL-1 protocol, the SPID's (Service Profile Identification Number) as well as the directory numbers (DN's) must be programmed in the terminal. SPID's are numbers that the phone company assigns, which are associated with each of the line's two channels. If the SPID's are not configured correctly, the terminal will not work.

E3. Line characteristics for the USA. Line request model.

In the case of the USA, ISDN lines have many configurable parameters, such as the protocol used, number of channels, type of call the channels accept (analog and digital, only analog, etc.), etc.

To ensure that the line's characteristics are appropriate for use with the PHOENIX, we recommend you use the following model to request it:

LINE REQUEST MODEL:

We request an ISDN Basic Rate Interface (BRI) line for an Audio Codec unit.

We will provide the NT1 and need a U 2B1Q interface on a standard RJ-11 jack.

We support only *National ISDN-1 Protocol*.

We need clear 64 Kbps access. (56 Kbps is not acceptable)

Long distance prefix (10XXX) must be enabled.

Long distance carriers: MCI, AT&T, Sprint

If you use IOC Capability Packages, please use *Bellcore Capability Package M*. If you do not use IOCs, use the following information:

- Two B channels.
- Circuit Switched data (CSD) and Circuit Switched Voice (CSV) on both channels
- Dynamic TEI assignment.
- No features or special services such as packed mode data on D channel, CACH EKTS, call forwarding, hunt groups, etc..

Please provide:

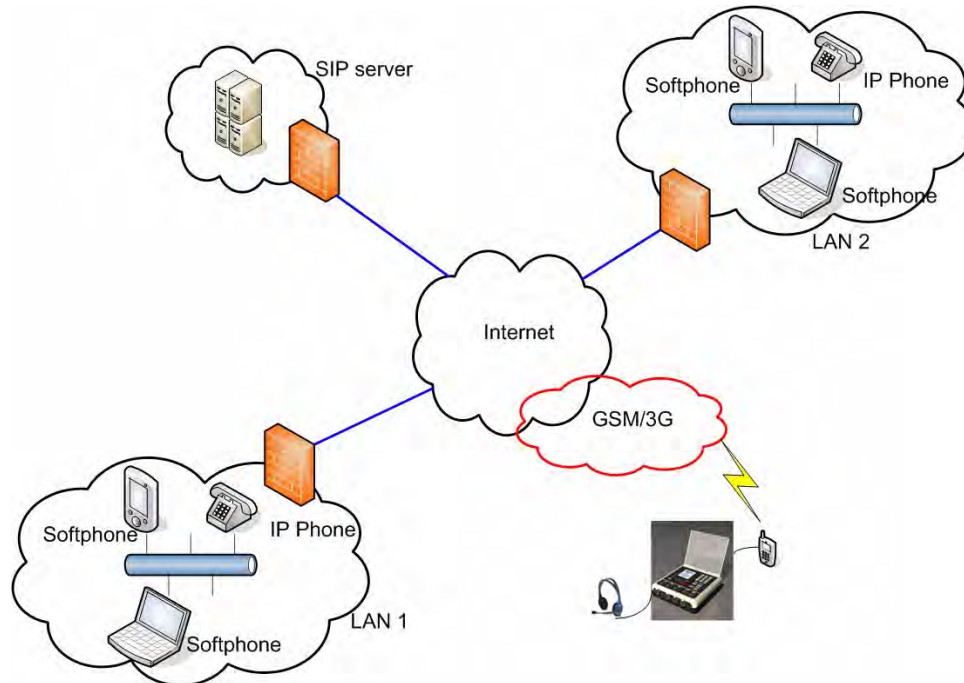
- Two SPID numbers.
- Two DNs

E4. Connecting the PHOENIX MOBILE to the ISDN line.

If the U interface is to be used (more common in the USA), the line will be connected to the RJ-11 jack (6 pins) of the ISDN module using a 6-pin modular telephone cable (RJ-11 to RJ-11). Note that the connection point to the U interface provided by the phone company may be a 6-pin RJ-11 type jack or an 8-pin RJ-45 type (the standard allows both types). If the jack provided is an RJ-45 type, the RJ-11 connector (6 pins) on the cable may be inserted into the center of the RJ-45 jack (8 pins).

ANNEXE F: 3G and GSM Communications.

This annexe refers to USB communication interface of PHOENIX MOBILE. 3G is commonly used as a general word used to signify data transmission through a mobile link, like the one provided by a 3G modem or a mobile phone. This type of communication could be described, in very simple words, as an IP connection that uses a 3G link to the network. It refers to the technology used to access the corresponding network, in this case, the Internet.



General scenario for data transmission

In order to establish the connection, we will be using the mobile phone as a dial-up modem. A pre-configured (standardised) number will be dialed, and certain identification parameters (username and password) will be given to the server using PPP protocol.

GSM (Global System for Mobile communications) is the main mobile communication standard used in Europe. The frequency bands in Europe are 900MHz and 1800MHz, and in USA are 850MHz and 1900MHz. A tri-band phone can be used in Europe as well as in the States. Although most GSM phones are used for voice communication, they can also be used for data transmission or for Internet access via GPRS.

CDMA (Code Division Multiple Access) is the mobile communication standard used in USA and Asia. European countries have not deployed CDMA networks.

Data transmission technologies:

- GSM family.
 - o **W-CDMA (3G)**. This refers to the radio technology of **UMTS**. At the end of 2007, W-CDMA represented over 70% of commercial 3G networks.
 - o HSPA (3.5G). High Speed Packet Access is currently the fastest access technology for data communications, with HSDPA (High Speed Downlink Packet Access) providing 14.4 Mbps on the downlink and HSUPA (High Speed Uplink Packet Access) returning 5.8 Mbps on the uplink. Not all operators have HSDPA deployed in all areas.
- CDMA family.
 - o **EV-DO** (EVolution Data Optimised). CDMA2000 is the dominant wireless technology in USA. CDMA2000 usually refers to voice traffic, whereas (CDMA) EV-DO usually designates data traffic.