



AEQ PHOENIX STRATOS



USER'S MANUAL

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Firmware Versions: Micro 5.20 / DSP 3.33 / FPGA 3.04 or higher
ISDN Module: Micro 1.01 / TA 6.00

Software Version: AEQ Control/PHOENIX 2.2.0.4 or higher

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1. INTRODUCTION.

1.1. General description.

AEQ PHOENIX STRATOS is a multiformat, multi-algorithm dual channel audiocodec designed for stationary rack-mounted applications, like links between studios, STL or mobile equipment connections. Each one of the two channels is stereo, so actually the unit houses two stereo codecs with independent AES/EBU digital and analog inputs and outputs.

PHOENIX STRATOS is a rackmounted audiocodec offering the options of working over ISDN, V35/X21 and IP networks. When used over the IP communications interface, it is fully compatible with the EBU-TECH 3326 technical specification issued by the EBU N/ACIP working group. This specification was developed to guarantee compatibility between devices made by different manufacturers for professional quality audio contribution applications over IP networks.

PHOENIX STRATOS enables its users to work simultaneously over two channels with professional quality in independent dual mono format or in stereo, or combination of both to several different destinations.

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

We recommend you reading this manual and “AEQ ControlPHOENIX” application user’s manual carefully and completely before using the unit, although chapter 5 offers a quickstart user’s guide.

1.2. Functional specifications.

- Management of two full-duplex channels (dual mono or stereo) over several possible communication interfaces: dual IP, V.35/X21 and ISDN.
- Inputs:
 - Dual Stereo analog audio input 4 x XLR female. 9Kohm. Electronic balancing. Line level.
 - Digital audio input. DB15 connector. Dual AES/EBU interface with SRC
- Outputs:
 - 2 stereo analog audio outputs. 4 x XLR male. Output impedance <100 ohm. Electronic balancing. Line level.
 - Digital audio outputs. Two digital AES/EBU audio outputs. It is possible to synchronize the output with the sampling frequency of the input.
 - Headset output. 1 x ¼” stereo jack with volume control on front panel.
- Synchronism: each output extracts it from the channel or alternatively its AES/EBU input.
- IP Interface. Ethernet Port. Audio and control interface over IP. LAN 10/100 base T RJ45 connector.
- PHOENIX STRATOS implements SIPv2 for the IP interface according to RFC-3261 and RFC-3264.
- **AEQ offers the “SmartRTP” call initiation protocol in order to greatly simplify the operation of the unit.**
- **AEQ also offers the services of its 2 own SIP servers (one of them working as main server and the other one as backup server) at no additional cost.** More information in section **ANNEXE C.**
- V.35 interface over DB25 connector.

1.3. Available encoding modes.

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

We recommend consulting **ANNEXE A** to see a detailed description, as well as additional information on these encoding modes.

- OPUS
 - Mode: Mono / Stereo
 - Bit rate: 12Kbps / 20Kbps / 32Kbps / 64Kbps / 128Kbps / 192Kbps
 - Sampling rate: 48KHz
- G.711 A-Law or μ -Law mono
 - Mode: Mono
 - Bit rate: 64Kbps
- G.722
 - Mode: Mono
 - Bit rate: 64Kbps
- AEQ LD+ *
 - Mode: Mono / Stereo
 - Bit rate: 64Kbps / 128Kbps / 192Kbps / 256Kbps / 384Kbps
 - Sampling rate: 16KHz / 32KHz / 48KHz
- MPEG-1/2 LII (MP2) ***
 - Mode: Mono / Stereo / Dual / JStereo
 - Bit rate: 64Kbps / 128Kbps / 192Kbps / 256Kbps / 384Kbps
 - Sampling rate: 16KHz / 24KHz / 32KHz / 48KHz
- AAC-LC ****
 - Mode: Mono / Stereo / MS Stereo
 - Bit rate: 32Kbps / 64Kbps / 96Kbps / 128Kbps / 192Kbps / 256Kbps
 - Sampling rate: 24KHz / 32KHz / 48KHz
- AAC-LD ****
 - Mode: Mono / Stereo / MS Stereo
 - Bit rate: 32Kbps / 64Kbps / 96Kbps / 128Kbps / 192Kbps / 256Kbps
 - Sampling rate: 48KHz
- PCM
 - Mode: Mono / Stereo
 - Bit rate: between 384 and 2304Kbps
 - Sampling rate: 32KHz / 48KHz
 - Bits per sample: 12 (DAT) / 16 / 20 / 24
- Hybrid mode 1
 - TX: MPEG1-LII_64_48
 - RX: G722
- Hybrid mode 2
 - TX: G722
 - RX: MPEG1-LII_64_48
- AUTOSYNC (only for ISDN and V35)

* Proprietary mode, exclusive for AEQ Phoenix, which combines very high quality and low delay.

** Only the 64 and 128Kbps modes are available for **ISDN** communications and all the 64, 128 and 256Kbps modes for **V35**.

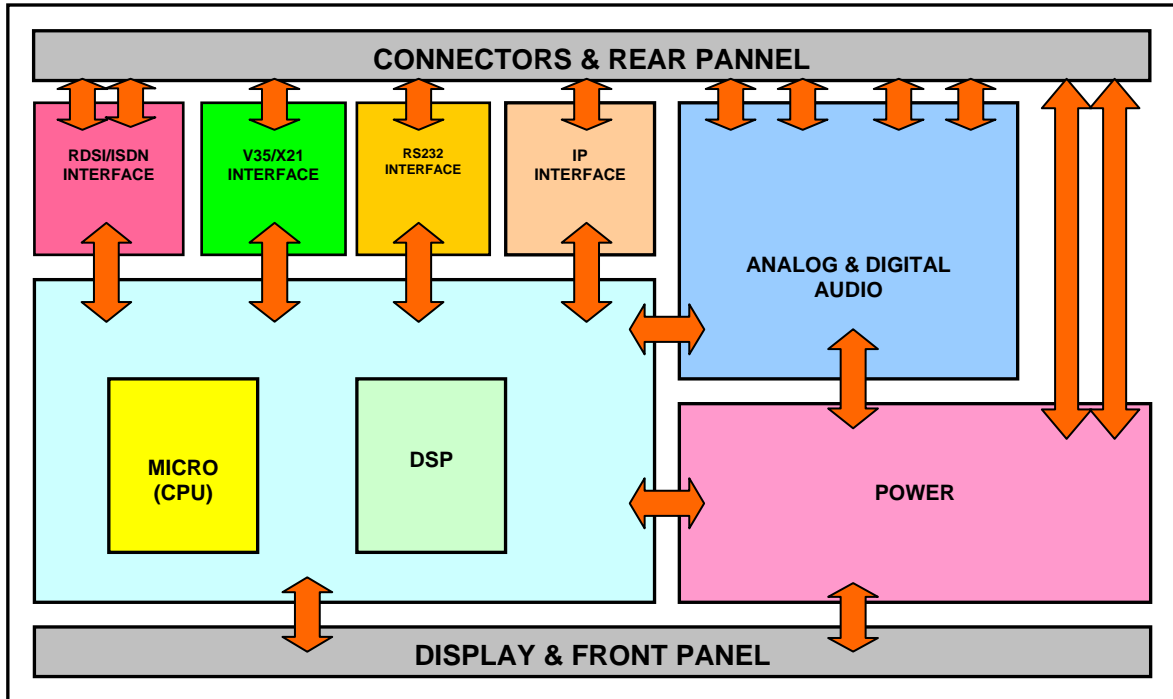
*** The 64 and 128Kbps modes over **ISDN** interface include the option of carrying a **continuous data channel** at 9600 bps.

**** **AAC-LC** and **AAC-LD** are **optional** modes. For more information, please consult AEQ Sales Department or authorized AEQ distributors.

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

1.4. Block diagram.

The internal design of the unit is organized in various functional modules that work together to make the Phoenix STRATOS audiocodec a complete communications platform. The block diagram below gives a simplified view of the distribution of these modules on the mainboard of the equipment, as well as a brief description of the associated functions.



Phoenix STRATOS internal distribution detail

- The audio board contains all the audio input and output electronics, both analog and digital, as well as the V.35/X21, IP & ISDN communication interfaces. It also includes a serial port for high level debugging applications and for connecting other equipment for carrying auxiliary data.
- The power supply module is responsible for converting the AC input into the various DC voltages needed inside the device.
- The board containing the FPGA electronics also provides the interface toward the microprocessor modules (CPU) and DSP, all of which form the processing and real control core of the unit.
- The front panel consists of an OLED display showing the control and configuration menu as well as the set of buttons and indicators for operating the equipment, including the alphanumeric keypad and the VU meters.

1.5. Compatibility with other AEQ codecs.

PHOENIX STRATOS offers its users the option of connecting with other AEQ equipment.

It's compatible with Phoenix ALIO, MERCURY, STUDIO, VENUS, VENUS V2, MOBILE, LITE, POCKET and PC in IP communications. The "**SmartRTP**" mode and **OPUS** encoding algorithm can be used with Phoenix ALIO, MERCURY, STUDIO, VENUS and VENUS V2 updated to CPU firmware version 5.20 or above.

It's compatible with Phoenix MOBILE and STUDIO in ISDN communications.

The ISDN interface allows its users to communicate with other devices equipped with ISDN interfaces, included in TLE02D, SWING, MPAC, EAGLE and COURSE ISDN boards.

V35 interface is compatible with EAGLE and Ranger devices.

1.6. Compatibility with other manufacturers' codecs.

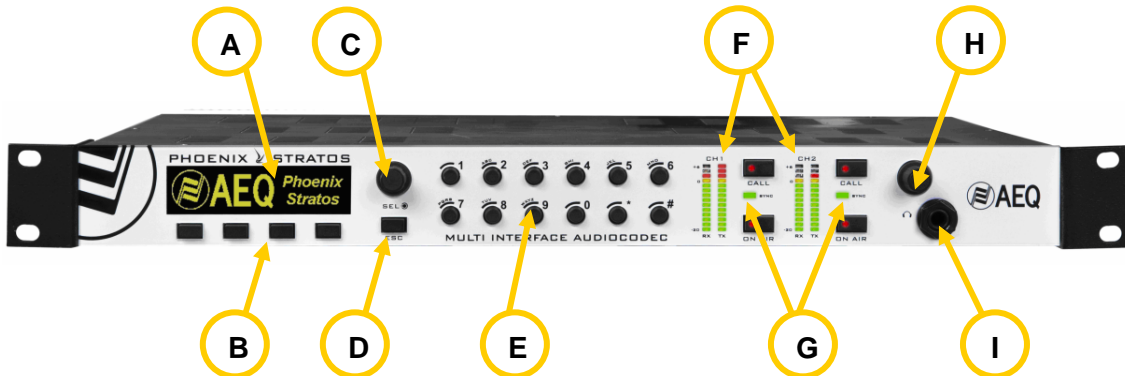
PHOENIX STRATOS is a stationary encoder over IP, fully compatible with the EBU-TECH 3326 technical specification formulated by the EBU N/ACIP working group. The specification was developed to guarantee compatibility between different manufacturers in applications for contributing professional quality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

For ISDN communications compatibility is usual with almost all equipments, although it will depend essentially on the coding algorithms implemented by the manufacturers of the other equipment, since the signalling part is practically universal. This is why we recommend consulting, before operating with other manufacturers' equipment, the technical characteristics of that equipment.

2. PHYSICAL DESCRIPTION OF THE UNIT.

To understand how the PHOENIX STRATOS 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 controls.



2.1.1. Multifunction display. A

High-contrast 256x64-pixel OLED screen in black and yellow. Within the internal menu, the screen gives a generic display of three differentiated areas:

- The upper area indicates the name of the menu where you will find the currently displayed options.
- The central area is associated with the various functions in whatever menu is currently being displayed; it is also the area where notifications and messages are shown.
- The lower screen area always shows options associated with the four lower keys that function as contextual buttons.

2.1.2. Screen-associated keys / Contextual buttons. B

Four multifunction buttons. Contextual operation depending on the menu options being displayed on the last line in the multifunction screen at any moment.

2.1.3. Browsing encoder. C

This enables cursor browsing in the various multifunction screen menus. Turning it makes the selection on the cursor or highlighted on the screen of the options shown on the various lines of the display, shifting from up to down and vice versa. Turning the encoder to the right raises the cursor or highlighted element; turning it to the left lowers it.

Pressing the button allows you to validate the selection you currently have active on the cursor or highlighted on the screen.

2.1.4. ESC button. D

Performs the Escape function, enabling you to shift backward in the various multifunction screen menus or cancel selections. Its operation produces the opposite effect of pressing the browsing encoder; it returns you, in the multifunction screen, to the menu immediately previous to the one currently being displayed.

RECOMMENDATION: by repeatedly pressing this key, a relatively untrained user will always return to the initial system screen, regardless of the internal menu being displayed immediately before pressing; no changes will be saved.

2.1.5. Alphanumeric keypad. **E**

Standard 12-keys alphanumeric keypad that enables the user to perform many functions, including the following:

- Enter a number when you wish to make a call through the ISDN interface
- Enter numbers and letters when you want to make a call through the IP interface, or when you create or edit phone book entries.

To change between different letters of the alphabet associated with the same key, press the key repeatedly until the desired letter appears, just as you would with a cell phone.

NOTE: The * key lets you switch between lower and upper case and only numbers. The 1 and # keys allow you to create the special characters (among there are @ and ".") and the 0 key allows you to create the number "0" and the space character.

2.1.6. VU meters. **F**

Two stereo monitoring VU meters. They can be configured in 3 different modes: as stereo input signal & stereo output signal for each one of the two channels, or as two independent mono signals with their corresponding input and output levels (both channels simultaneously).

The serigraphed lettering on the front panel identifies them, from left to right, as "CH1 RX", "CH1 TX", "CH2 RX" and "CH2 TX". If a stereo connection is made, and vu meters are configured this way, "CH1 RX" and "CH1 TX" represent the L and R transmitted signals, while "CH2 RX" and "CH2 TX" correspond to the L and R received signals, corresponding to channel 1 or 2 depending on the vu meters mode selected. For further information, see section "3.3. VUMETERS menu" of this manual.

Each of the two VU meters is composed of 14 LEDs, of which the top three are red, the fourth yellow (representing +0dB) and the rest green. Each one of the 14 LEDs represents 2dB; the complete scale of each VU meter ranges from +6dB to -20dB.

2.1.7. Communication status control information. **G**

There are two identical sections offering communication status information; each is associated with a different communication channel (1 or 2). The buttons on the left correspond to communication channel 1 and the buttons on the right correspond to communication channel 2.

Each section is composed of three elements.

- **CALL** button: pressing it allows you to make a call. For further information, see the QUICK OPERATING GUIDE section of this manual.
- **SYNC** LED: to indicate the connection status. This LED has three statuses:
 - Off: no connection.
 - Red: trying to synchronize.
 - Green: connected and synchronized.
- **ON AIR** button: enables full duplex communication, connecting the signal from the selected input (analog or digital) with the output of the defined communication interface.

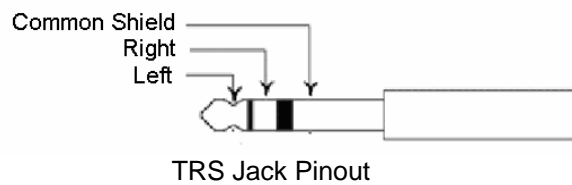
2.1.8. Headphones volume. **H**

Potentiometer to control the output volume in the monitoring headphones. When this knob is turned all the way to the left, the output audio level will be minimal or null; turning it all the way to the right will produce the maximum output level. All the intermediate positions are possible.

NOTE: We recommend prudence when adjusting the volume, since excessively high levels can be harmful to the listener’s health.

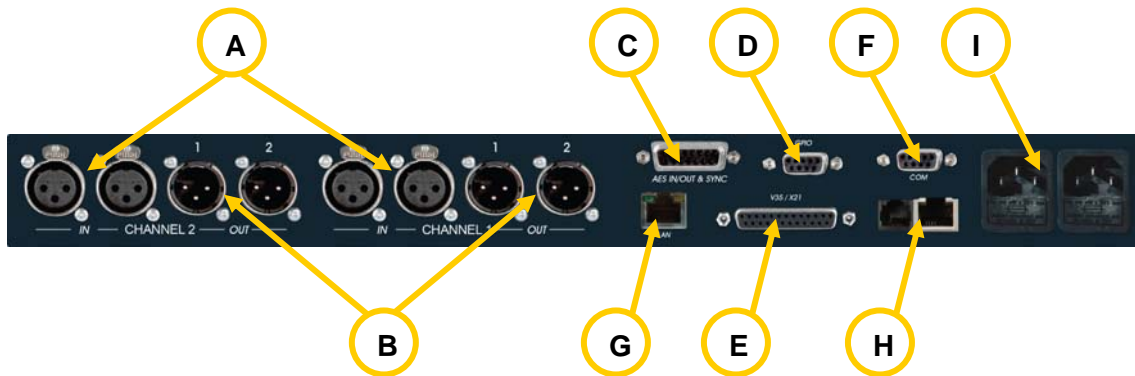
2.1.9. Monitoring headphones output. **I**

¼” TRS Jack. Asymmetrical connection.



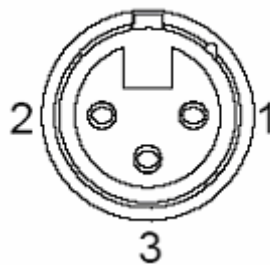
Configuring the codec by using the menu in the front panel display will let you associate the signals to be monitored independently for the left and right headphone. See section “**3.2. HPHONES menu**”.

2.2. Description of the rear panel and connections.



2.2.1. Analog line stereo inputs. **A**

XLN - 3p female connector. Balanced connection. Two pair of connectors L (1) and R (2) or independent mono channels (the left connector corresponds to L input and the right one corresponds to R input).

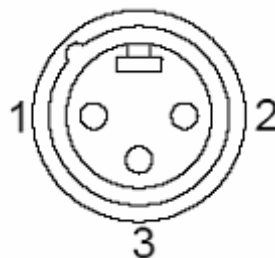


XLR 3p Female-panel Pinout

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

2.2.2. Analog line stereo outputs. **B**

XLN - 3p male connector. Balanced connection. Two pair of connectors L (1) and R (2) or independent mono channels (the left connector corresponds to L output and the right one corresponds to R output).

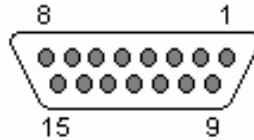


XLR 3p Male-panel Pinout

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

2.2.3. AES/EBU inputs/outputs. **C**

The physical connector used is a DB15 female, with the following pinout:



DB15 connector pinout identification

- | | |
|------------------------|-------------------------|
| - Pin 1: AES1_IN+ | - Pin 9: AES1_IN- |
| - Pin 2: AES2_IN+ | - Pin 10: AES2_IN- |
| - Pin 3: N/A | - Pin 11: AES1_OUT+ |
| - Pin 4: AES1_OUT- | - Pin 12: AES2_OUT+ |
| - Pin 5: AES2_OUT- | - Pin 13: N/A |
| - Pin 6: SYNC_IN_AES+ | - Pin 14: SYNC_IN_AES- |
| - Pin 7: SYNC_OUT_AES+ | - Pin 15: SYNC_OUT_AES- |
| - Pin 8: GND | |

Considerations:

- The signals containing audio and synchronizing information (Channel 1 and Channel 2) are compliant with the AES-3 standard. The signals that only contain synchronizing information (Synchronism) are governed by the AES-11 standard.
- The synchronizing input signal can be used to change the sampling rate of the AES/EBU output signal (set by default at 48 kHz).
- The synchronizing output signal will either repeat an external synchronizing input signal, or will take the internal 48 kHz signal to the synchronizing input of other AES/EBU equipment items. See menu description in section 3.1.3.3.
- When both of the two audio codec channels are used independently of each other, the audio carried by each of them will be output through the connector corresponding to channel 1 or 2.

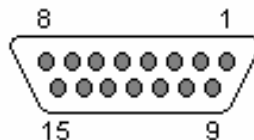
Digital and analog outputs are always simultaneously availables.

2.2.4. GPIO connector. **D**

General Purpose Inputs and Outputs connector. Configurable functionality. For further information, see section “3.1.8.4. GPIOs configuration” of this manual.

- NOTE:**
- The output signals are open collector active-low.
 - The input signals are opto-coupled with internal 2.2Kohms resistor. They are active-high.

The connector is a DB15 female type, with the following pinout:



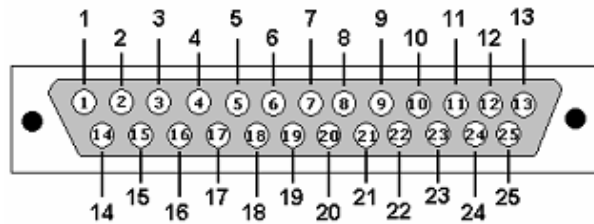
DB15 connector pinout identification

- | | |
|---------------|----------------|
| - Pin 1: GPO1 | - Pin 9: GPO2 |
| - Pin 2: GPO3 | - Pin 10: GPO4 |
| - Pin 3: GND | - Pin 11: N/A |
| - Pin 4: N/A | - Pin 12: N/A |
| - Pin 5: N/A | - Pin 13: N/A |
| - Pin 6: GPI1 | - Pin 14: GPI2 |
| - Pin 7: GPI3 | - Pin 15: GPI4 |
| - Pin 8: GND | |

2.2.5. Interfaz X.21/V.35.

E

DB25 physical connector.



DB25 connector pinout identification

Allows an external terminal adapter (TA) to be connected at speeds of 64/128/256 kbps. Two types of TA can be used:

- V35 (V11) interface: Uses unbalanced asynchronous control signals, while the synchronous clock and data signals are balanced.
- Interface which, at the physical level, is compatible with X.21 / RS422 protocol levels.

The pin-out detail and the signals available in the DB-25 connector are the following:

Physical Level	Signal	Pin
RS 422 Bal	Reloj (RCX) V+	13
RS 422 Bal	Reloj (RCX) V -	14
RS 422 Bal	TX V+	11
RS 422 Bal	TX V-	10
RS 422 Bal	RX V+	21
RS 422 Bal	RX V-	19
RS232Unbal	DTR (Data Terminal Ready)	20
RS232Unbal	DCD (Data Carrier Detect)	8
RS232Unbal	GND	7

Pins 4 and 5, equivalent to RS232 RTS and CTS signals, are strapped internally. The AEQ PHOENIX STRATOS can be configured to define whether the DCD signal will manage communication (DCD CONTROLLED) or, on the contrary, the connection will be without DCD management (PERMANENT). In the latter case, it will only be necessary to connect the six shaded signals to establish the connection to the terminal adapter.

CAUTION:
Disconnecting the V.35 interface while the equipment is powered up could cause internal damage.

To connect the AEQ PHOENIX STRATOS with an external V35 Terminal Adapter, you will need to search in the TA user's manual for the exact pinout matches.

Because the TA equipment currently on the market has a great variety of connectors, AEQ cannot provide details of all the possible connections. The following table shows, as an example, the connection to an ISO-2593 standard terminal adapter in an MRAC-34 connector.

CCITT	DESCRIPCIÓN	DTE	DB25		Terminal Adaptador Conector MRAC-34	DCE
102	SIGNAL GROUND	GND	7	→	B	GND
103	TRANSMITTED DATA A	TXD A	11	→	P	TXD A
	TRANSMITTED DATA B	TXD B	10	→	S	TXD B
104	RECEIVED DATA A	RXD A	21	→	R	RXD A
	RECEIVED DATA B	RXD B	19	→	T	RXD B
115	RX. CLOCK V+	RXC A	13	→	V	RXC A
	RX. CLOCK V-	RXC B	14	→	X	RXC B

Some external terminal adapters are configurable. In this case, you will need to ensure that they have these parameters:

- 64kbps, 128kbps or 256kbps
- Synchronous
- CD Forced (ON)
- Ignore DTR

2.2.6. COM connector. **F**

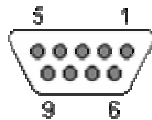
In IP mode, allows you to connect an external equipment (a PC, for instance) in order to transmit and receive auxiliary data in a transparent way. An ancillary data channel at 9600, 19200 or 38400 bauds is supported that is transmitted over IP, independently of the audio stream. Only one auxiliary data stream can be used, so it has to be assigned to one of both codec channels.

In ISDN mode, a data channel at 9600 bps is supported in some MPEG-2 encoding modes.

Data transmission is performed according to RS232 protocol on a DB9 female connector. A male-female cable must be used to connect the codec to a PC.

The active pins at the port are:

PHOENIX		PC
TX 2	→	2
RX 3	←	3
GND 5	--	5



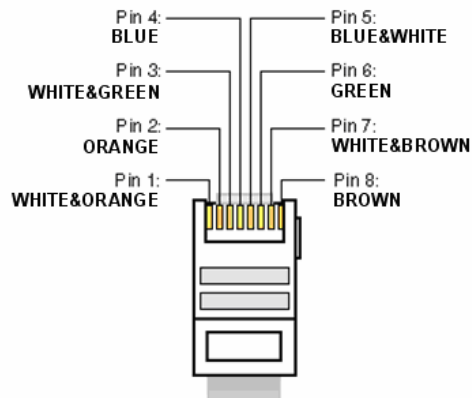
Note that hardware flow control must be disabled at the PC

2.2.7. Ethernet port. **G**

PHOENIX STRATOS is equipped with an Ethernet port. 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 is also used to configure and manage the unit from one or several PCs using the remote control software.

Please consult "AEQ ControlPHOENIX" application manual.

Physically, the connector is RJ45 10/100 BT, with the following pinout.



RJ45 connector pinout

2.2.8. ISDN port.

H

According to the UIT-T, the **Integrated Services Digital Network (ISDN)** can be defined as a network that has evolved from the Integrated Digital Network (IDN), and which provides end-to-end digital connections to deliver a wide range of services, both voice and other types, and which users access through a set of standardized interfaces.

It can thus be said that it is a network that is an outgrowth of the existing telephone network and that, because it offers end-to-end digital connections, enables the integration of a multitude of services in a single access, regardless of the nature of the information to be transmitted and of the terminal device that generates it.

The connector employed is a standard RJ45 for the S interface and a standard RJ11 for the U interface.

2.2.9. Power supply.

I

Mains input connector(s), including fuse and fuseholder.

Power supply: universal 90 - 240 V AC, 50-60Hz. 15 VA. IEC 3 pin connectors

The unit can include two internal power supplies, one of them working as redundant of the other one.

3. INTERNAL MENU DESCRIPTION. MANUAL CONTROL.

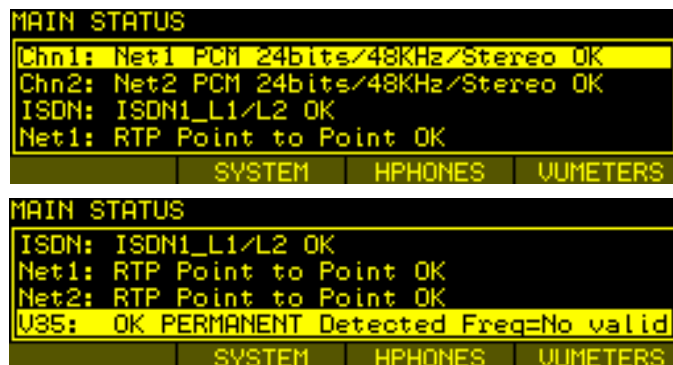
The configuration and operation of the PHOENIX STRATOS unit are accomplished **locally** on the OLED screen in the front panel or **remotely** by means of the application “**AEQ ControlPHOENIX**” (AEQ Phoenix STRATOS, ALIO, STUDIO, MERCURY, VENUS and VENUS V2 Audiocoders Configuration and Control Software). In this chapter first option is described. The OLED screen presents all the information and menus associated with the operating mode of the audiocodex.

Initially, when the Phoenix STRATOS is powered up, for 5 seconds a welcome screen with the AEQ logo and the name of the device is displayed.



Welcome screen

After the welcome screen, the display will show the MAIN STATUS screen indicating the configuration of the various communication interfaces. In order to see all the interfaces options you have to scroll by means of rotary encoder. The right part of the screen shows the position where you are by means of a scroll line; this function works in every screen where all the information is not shown in just one view.



MAIN STATUS screen detail

- **Chn1:** indicates the communication interface with which communication channel number 1 is associated and the encoding algorithm used in the connection, among many other details.
- **Chn2:** indicates the communication interface with which communication channel number 2 is associated and the encoding algorithm used in communication, among many other details. Exact same configuration as for Chn1.
- **Net1:** shows a summary of the configuration and status of the IP interface over the Net1 channel: PROXY SIP CONNECTED for instance.
- **Net2:** shows a summary of the configuration and status of the IP interface over the Net2 channel: RTP Point to Point for instance.
- **V35:** shows the status of the V35 communications interface: “OK PERMANENT Detected Freq=No valid” for instance.

The last line of this MAIN STATUS screen shows three options associated with the contextual buttons located beneath the OLED display.

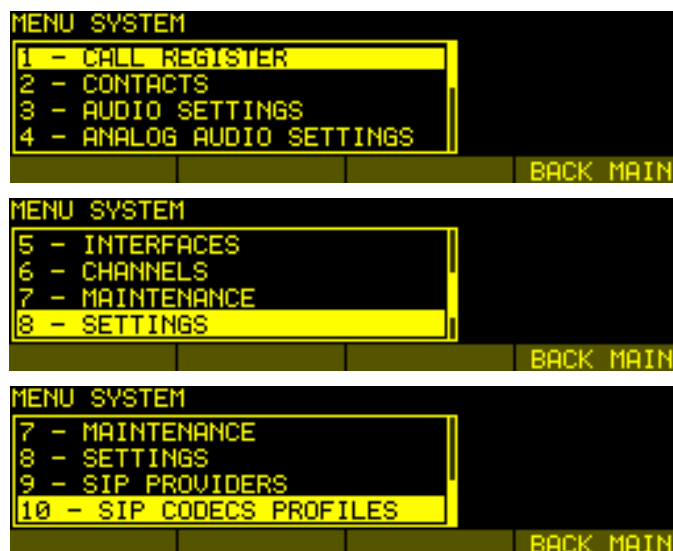
- **SYSTEM:** internal control, configuration and operating menu.
- **HPHONES:** configuration of the headphone output for control monitoring, available on the front panel of the unit.
- **VUMETERS:** configuration of the working mode of the two groups of VU meters available in the front panel.

3.1. SYSTEM menu.

The SYSTEM menu is the internal control, configuration and operating menu for the PHOENIX STRATOS unit.

The internal SYSTEM menu is organized logically:

1. **CALL REGISTER:** information regarding calls sent or received.
2. **CONTACTS:** Internal phone book listing contacts. Its maximum capacity is 255 records.
3. **AUDIO SETTING:** Configuration of the audio signals inside the unit and its association to the communication channels.
4. **ANALOG AUDIO SETTINGS:** Specific configuration of the analog audio inputs and outputs.
5. **INTERFACES:** Configuration of communication interfaces and modules.
6. **CHANNELS:** Configuration of the communication channels, indicating the used communication interface, the applied encoding algorithm and other options.
7. **MAINTENANCE:** Internal maintenance menu for the codec.
8. **SETTINGS:** Local configuration of the unit, including the IP configuration of its Ethernet ports.
9. **SIP PROVIDERS:** Configuration of various Proxy SIP servers.
10. **SIP CODECS PROFILES:** Configuration of coding profiles for working over IP communications (just for SIP).



SYSTEM menu detail

You can shift among the various options by turning the rotary encoder. You can select an option by pressing on the rotary encoder.

To quit this menu and return to the immediately previous MAIN STATUS menu, simply press the ESC button or the button associated with the BACK MAIN option located in the lower right-hand corner of the OLED display.

3.1.1. CALL REGISTER menu.

Menu displaying information regarding sent or received calls in any communication interface. You can access the CALL REGISTER menu from the SYSTEM MENU, in the first option, by pressing the rotary encoder.



CALL REGISTER detail

The CALL REGISTER menu contains two submenus:

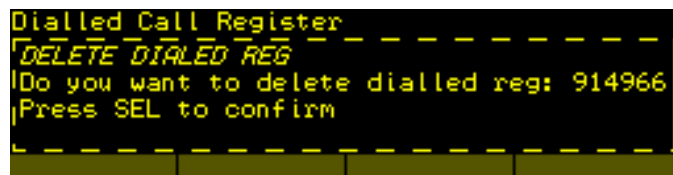
- **Dialed List:** List of the calls made by the unit. Stores the information associated with the last 10 calls made.
- **Received List:** List of the calls received by the unit. Stores the information regarding the last 10 calls received.



DIALED CALL REGISTER list

You can shift among the various list entries by turning the rotary encoder. By pressing on the rotary encoder you can okay an option and access to more details regarding that call.

You can delete one of the records stored in these called lists, or all of the records, by using the DEL (delete the selected list entry highlighted in yellow) and DEL ALL (erase all the entries) options in the contextual buttons. The unit will request confirmation of this action.



DEL confirmation detail

To quit this menu and return to the immediately previous one, simply press the ESC button.

Each one of the calls made or received generates a calling or called contact record, as well as the date of the call. By pressing the rotatory encoder over an specific call you can access to details screen.

To make a call to the selected contact of the list, you have to press the CALL button corresponding to the channel you want to use, and then press the DIAL contextual key.



Detail of the DIALED CALL REGISTER entries

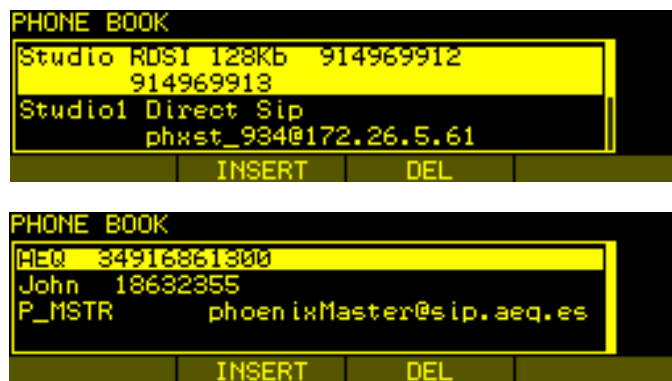
- **NUMBER / URI:** number or IP identifier associated with the contact.
- **CHANNEL:** communication channel (1 or 2) over which the connection was made.
- **Start:** date and time of the beginning of the communication.
- **End:** date and time of the end of the communication.
- **INTERFACE:** interface (IP, ISDN or V35) over which the connection was made.

To quit this menu and return to the immediately previous one, simply press the ESC button.

3.1.2. CONTACTS menu.

Contacts phone book menu. You can access the CONTACTS menu from the SYSTEM MENU, in the second option, by pressing the rotary encoder.

The menu shows the internal contacts phone book in the unit, which can store up to 255 different records in alphabetical order.



Phonebook detail

Each of the stored entries includes a NAME field (contact name), two optional ISDN telephone numbers fields and an IP identifier field, also optional, which syntax depends on the type of established IP connection.

You can shift among the various list entries by turning the rotary encoder. You can okay an option by pressing on the rotary encoder.

There are two options displayed in the lower part of the screen associated with the contextual buttons located beneath the OLED display.

- INSERT.
- DEL.

To quit this menu and return to the immediately previous SYSTEM MENU, simply press the ESC button.

3.1.2.1. INSERT.

The INSERT option associated with the second contextual button allows you to create a new entry in the phone book.

The first part in the task of creating a new phone book entry is filling the NAME field (contact name), an alphanumeric identifier of the mnemonic type that needs to be assigned to the contact.

To compose the identifier, you will use the buttons on the front panel of the unit where, beneath the digit represented, you will find the letters allowed. Its operation is identical to that of a cell phone keypad; you'll need to press more than once to shift from one letter to another within the same button.

- The 1 key enables you to generate the characters: . , _ @ : % 1
- The 2 key enables you to generate the characters: a b c 2
- The 3 key enables you to generate the characters: d e f 3
- The 4 key enables you to generate the characters: g h i 4
- The 5 key enables you to generate the characters: j k l 5
- The 6 key enables you to generate the characters: m n o 6
- The 7 key enables you to generate the characters: p q r s 7
- The 8 key enables you to generate the characters: t u v 8
- The 9 key enables you to generate the characters: w x y z 9
- The 0 key enables you to generate the space character and 0.
- The * key lets you switch between lower and upper case and only numbers.
- The # key enables you to generate the characters: . , _ @ #

While you are creating any name, number or IP identifier, you will be able to use four options associated with the four contextual buttons. These same options are repeated in numerous menus in the PHOENIX STRATOS with the same functionalities:

- **DEL:** deletes the character placed on the left side of the cursor (except when the cursor is placed at the beginning of the identifier: in that case it deletes the character placed on the right side of the cursor).
- **<:** lets you move the writing cursor to the left.
- **>:** lets you move the writing cursor to the right.
- **OK:** validates the identifier created.



Detail of the creation of the alphanumeric identifier in phonebook

By pressing the rotary encoder, you can also validate the identifier created, or you can turn it to perform the same functions as the < and > contextual buttons.

To quit this menu and return to the immediately previous one, simply press the ESC button. Depending on the time of its activation, it may be that no change made in the last access to this menu will be validated and stored in memory.

The second part in the task of creating a new phone book entry is the ISDN number or IP identifier that needs to be assigned to the contact. You will access automatically this second step immediately after validating the name.

First you can insert the ISDN number (“TLF”), then a second number (“TLF2”) for the case of an ISDN link with two B channels (a 128Kbps bit rate).

Finally, you can insert the IP identifier (“URI”): several syntax are allowed depending on the selected communication mode, following the standard rules.

- When the selected working mode for the contact is “**PROXY SIP**”, the valid formats are:
 - **<unit_name>**, “phstr_231” for instance.
 - **<unit_name>@<SIP_server_domain >**:
“phstr_231@sip.aeq.es” for instance.
 - **<unit_name>@<SIP_server_IP_address>**:
“phstr_231@232.168.1.2” for instance.
 - **<unit_name>@<SIP_server>:<Port >**, when the SIP port is not 5060 (the one by default in SIP SERVER mode): “phstr_231@sip.aeq.es:5061” for instance.

- When the selected working mode for the contact is “**DIRECT SIP**”, the valid formats are:
 - **<unit_name>@<unit_IP_address>**:
“phstr_231@172.26.5.57” for instance.
 - **<unit_name>@<unit_IP_address>:<Port>**, when the SIP port is not 5061 (the usual one in Direct SIP): “phstr_231@172.26.5.57:5061” for instance.
- When the selected working mode for the contact is “**RTP Point to Point**”, the only valid format is:
 - **<unit_IP_address>**: “172.26.33.28” for instance.

NOTE: Any of those fields can be left blank when the contact will not be used for IP communications.

In this second step, the alphanumeric keypad on the front panel of the PHOENIX STRATOS works only as a numeric keypad in order to insert the ISDN number/s and as an alphanumeric keypad in order to insert the IP identifier.

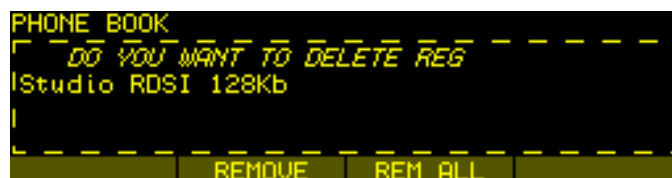


Detail of the record number/identifier creation screen

To quit this menu and return to the immediately previous one, simply press the ESC button. Depending on the time of its activation, it may be that no change made in the last access to this menu will be validated and stored in memory.

3.1.2.2. DEL.

The DEL option lets you eliminate one or all of the phone book records stored in the internal memory of the unit.



DEL options

After pressing the contextual button associated with the DEL option, a screen appears in which confirmation is requested and various options are offered:

- **REMOVE:** only the phone book entry selected in the previous screen when DEL was pressed will be deleted.
- **REM ALL:** all of the entries stored in the phone book will be deleted. The unit will request confirmation of this action.

By pressing the ESC button you can return to the previous menu, with no erasure or modification of the previously stored data.

3.1.2.3. Editing a contact.

While editing does not expressly appear as an option on screen, you can edit a previously stored phone book record simply by selecting the entry and pressing the rotary encoder.

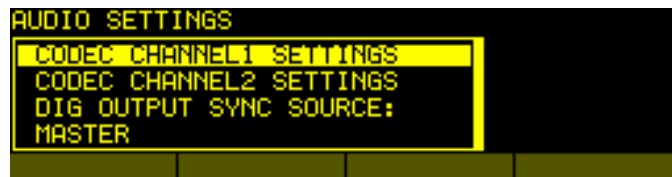
The procedure is identical to the one described above in the INSERT section.

3.1.3. AUDIO SETTINGS menu.

You can access the AUDIO SETTINGS menu from the SYSTEM MENU, in the third option, by pressing the rotary encoder. From this menu you can configure the routing of audio signals inside the Phoenix STRATOS, as well as the digital inputs and outputs and the synchronism.

The unit has 2 communications channels and each one has 2 Inputs and 2 Outputs in analog or digital format that can be configured as mono or stereo signals.

The specific adjustment options for analog inputs and outputs can be configured in the fourth option of SYSTEM MENU “ANALOG AUDIO SETTINGS” and the selection between analog or digital inputs for each audio channel can be done in the Main Menu: “MAIN STATUS / CHx STATUS / AUDIO IN SOURCE”.



AUDIO SETTINGS menu options

It is internally divided into a series of submenus.

- **CODEC CHANNEL 1 SETTINGS.**
- **CODEC CHANNEL 2 SETTINGS.**
- **DIG OUT SYNC SOURCE.**

To quit this menu and return to the immediately previous one, simply press the ESC button.

3.1.3.1. CODEC CHANNEL 1 SETTINGS menu.



AUDIO CODEC CHANNEL 1 SETTINGS menu detail

Enables the configuration of the audio signal associated with communications channel number 1, according to the following list of options:

- **CODEC INPUT MODE:** enables you to select the audio input between MONO (L, R or L+R) and STEREO MODE. This selection will be applied to both analog and digital input.



AUDIO INCODEC INPUT MODE options detail

- **LOCK L&R IN GAIN:** when activated (YES), the gain changes will be applied simultaneously to both channels of the encoder stereo input signal (whether it's analog or digital). In the other case, the changes will affect separately to L and R channels.



LOCK L&R IN GAIN menu detail

- **CODEC INPUT GAIN L (dB):** allows you to modify the left encoder input attenuation. This attenuation will be digitally implemented, but it will **affect both analog and digital inputs** (see section 3.1.6). Values in dB. 1 dB steps. Maximum value: 0dB (no attenuation); minimum value: -40dB (attenuation). When LOCK L&R IN GAIN is activated (YES), the gain adjustment will affect simultaneously the right encoder input channel.



CODEC INPUT GAIN L menu detail

- **CODEC INPUT GAIN R (dB):** the same as CODEC INPUT GAIN L (dB) for the right channel. When LOCK L&R IN GAIN is activated (YES), the gain adjustment will affect simultaneously the left channel.



CODEC INPUT GAIN R menu detail

- **LOCK L&R OUT GAIN:** when activated (YES), the gain changes will be applied equally to both channels of the encoder stereo output signal. In the other case, the changes will affect separately to L and R channels.



LOCK L&R OUTPUT GAIN menu detail

- **CODEC OUTPUT GAIN L (dB):** allows you to modify the left encoder output attenuation. This attenuation will be digitally implemented, but it will **affect both analog (see section 3.1.4) and digital outputs (always active)**. Values in dB. 1 dB steps. Maximum value: 0dB (no attenuation); minimum value: -40dB (attenuation). When LOCK L&R OUT GAIN is activated (YES), the gain adjustment will affect simultaneously the right channel.



CODEC OUTPUT GAIN L menu detail

- **CODEC OUTPUT GAIN R (dB):** the same as CODEC OUTPUT GAIN L (dB) for the right encoder output channel. When LOCK L&R OUT GAIN is activated (YES), the gain adjustment will affect simultaneously the left channel.



CODEC OUTPUT GAIN R menu detail

3.1.3.2. CODEC CHANNEL 2 SETTINGS menu.

This menu allows you to configure the audio signal associated with communications channel number 2, according to a list of options identical to the one defined in the previous section.



CODEC CHANNEL 2 SETTINGS menu detail

3.1.3.3. DIG OUT SYNC SOURCE menu.

This menu enables you to define the reference synchronizing signal for the digital audio outputs available in the DB15 AES IN/OUT & SYNC connector in the rear part of the unit. It can be configured as MASTER (uses the internal synchronism of the equipment) or SLAVE (uses external synchronism input SYNC_IN). This adjustment makes no difference when analog outputs are used.



DIG OUT SYNC SOURCE menu detail

3.1.4. ANALOG AUDIO SETTINGS menu.

You can access this menu from the SYSTEM MENU, in the fourth option, by pressing the rotary encoder. By contrast with the AUDIO SETTINGS menu, where you configure most of the Phoenix STRATOS AES/EBU digital audio part, this ANALOG AUDIO SETTINGS menu enables you to adjust all the parameters associated with the analog format audio output part.



ANALOG AUDIO SETTINGS menu detail

- **ANALOG OUTPUT 1 L ATT (dB):** allows you to modify the output attenuation of analog signal 1 L. Values in dB. 1 dB steps. Maximum value: 20dB, minimum: 0dB. This adjustment doesn't affect digital outputs.



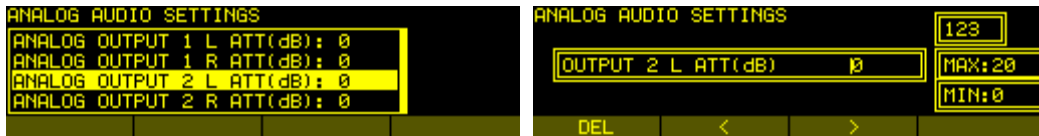
ANALOG OUTPUT 1 L ATT menu options

- **ANALOG OUTPUT 1 R ATT (dB):** allows you to modify the output attenuation of analog signal 1R. Values in dB. 1 dB steps. Maximum value: 20dB, minimum: 0dB. This adjustment doesn't affect digital outputs.



ANALOG OUTPUT 1 R ATT menu options

- **ANALOG OUTPUT 2 L ATT (dB):** allows you to modify the output attenuation of analog signal 1 L . Values in dB. 1 dB steps. Maximum value: 20dB, minimum: 0dB. This adjustment doesn't affect digital outputs.



ANALOG OUTPUT 2 L ATT menu options

- **ANALOG OUTPUT 2 R ATT (dB):** allows you to modify the output attenuation of analog signal 1R. Values in dB. 1 dB steps. Maximum value: 20dB, minimum: 0dB. This adjustment doesn't affect digital outputs.

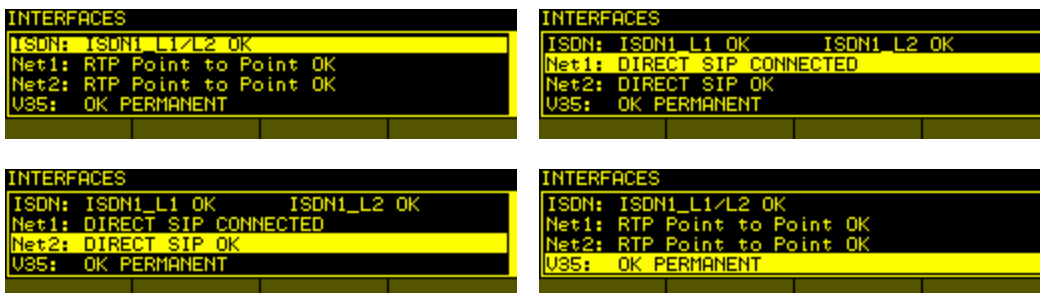


ANALOG OUTPUT 2 R ATT menu options

3.1.5. INTERFACES menu.

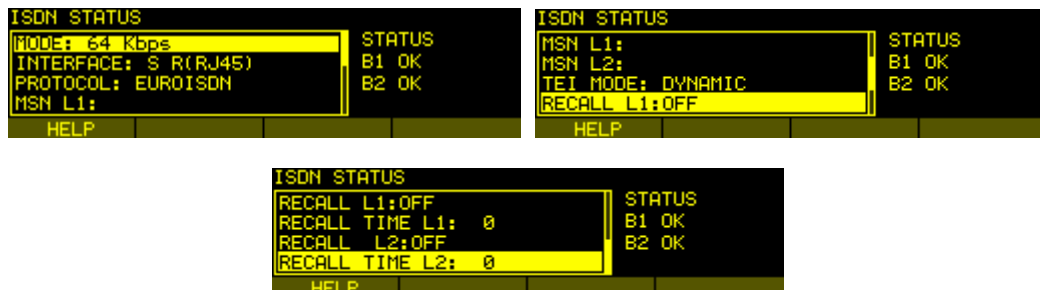
Menu for configuration of communication interfaces. You can access the INTERFACES menu from the SYSTEM MENU, in the fifth option, by pressing the rotary encoder.

The main screen of this menu displays the status information regarding the communication interfaces.



INTERFACES menu detail

Pressing on the rotary encoder will take you to the configuration menu for the currently highlighted communications interface (ISDN in the example):



ISDN interface configuration detail

3.1.5.1. ISDN interface.

When you press the rotary encoder when the ISDN option is selected, the ISDN STATUS submenu will appear and will enable you to completely configure the ISDN communication interface. The submenu offers these possibilities:

- **MODE:** lets you choose between working with two single 64Kbps B channels (that you can assign to different communication channels), or combine both channels in one 128Kbps channel (that will be used for just one communication channel).



Bit rate selection detail

- **INTERFACE:** allows you to choose between two connectors available on the SGA-03 module according to your geographic area and the ISDN protocol to be used: S (RJ45) or U (RJ11) connector.



ISDN physical interface selection detail

The U interface links the telephone exchange to the subscriber; it is a two-wire interface of the type used in analog telecommunications, and permits only a point-to-point connection (a single device connected to the interface).

The S/T interface is obtained through the U interface by passing through a Network Terminator (TR1) installed on the subscriber's premises; it is a four-wire interface that permits the multipoint connection of up to seven devices.

In USA, telephone companies do not supply the TR1; it is up to users to install these devices if they want them. The most normal situation is for the TR1 not to exist, and for the access to be accomplished through the U interface. In short, in USA the usual solution is access through the U interface unless the subscriber has requested or the telephone company has installed a TR1 of its own volition (the TR1 is normally a little box secured to the wall with a power supply and a luminous indicator of some type, into which the two-wire outside line enters, offering RJ-45 outputs).

- **PROTOCOL:** this submenu enables you to choose between the two different ISDN communication protocols available in the PHOENIX STRATOS, depending on your geographic area and the connector selected in the previous step.

The options are: EURO-ISDN and NATIONAL-1.



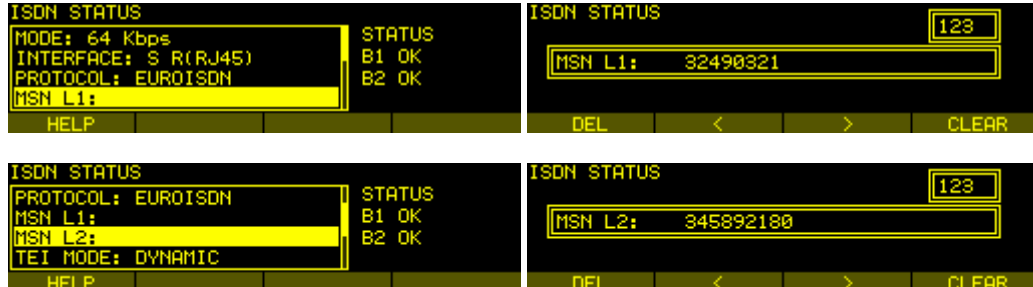
ISDN protocol options

In USA, despite the presence of several protocols that are incompatible with each other, there is, fortunately, one protocol that is relatively standardized (called National-1), and which is usually available in all the telephone exchanges.

Selecting NATIONAL-1 enables the SPID (Service Profile Identification Number) option in this same submenu. For further information, see ANNEXE B.

In Europe and other geographic areas there is a protocol called EURO ISDN. There are minor differences among the EURO ISDN standards applied in different countries. These differences do not affect the Phoenix STRATOS.

- **MSN L1 and MSN L2:** acronym for MULTIPLE SUBSCRIBER NUMBER, enables you to independently identify each of the B channels in the same ISDN network.



MSN L1 & L2 selection & creation menu detail

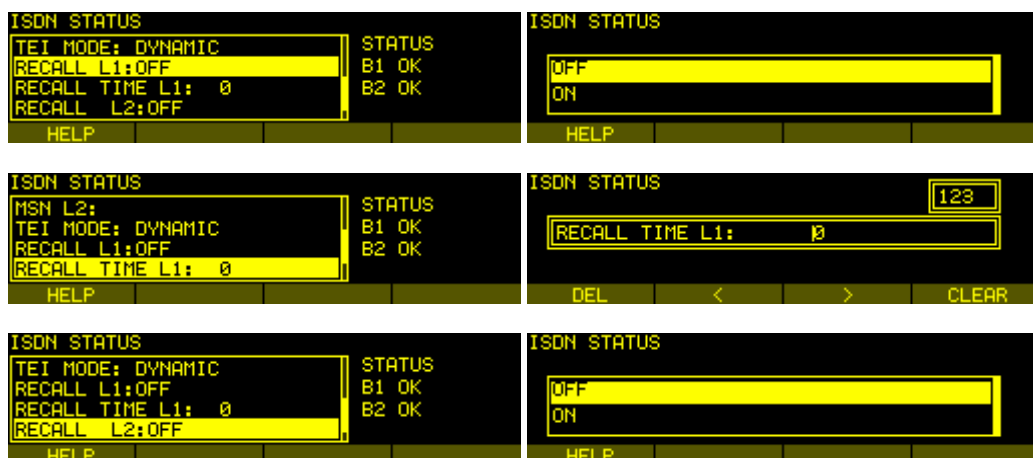
Like the SPID information, this MSN information must be furnished by the service provider where the ISDN line is connected. For further information on this subject, see ANNEXE B.

- **TEI MODE:** Terminal Equipment Identifier mode, which enables you to select your working mode, FIXED or DYNAMIC. The “DYNAMIC” option will usually be the right one, although some specific switchboards may require a FIXED working mode.



TEI configuration options

- **RECALL L1:** lets you activate/deactivate the recall option for B1 channel (whether it's associated to communication channel 1 or 2).
- **RECALL TIME L1:** lets you define the time interval associated with the recall option for B1 channel.
- **RECALL L2:** lets you activate/deactivate the recall option for B2 channel (whether it's associated to communication channel 1 or 2).
- **RECALL TIME L2:** lets you define the time interval associated with the recall option for B2 channel.





RECALL L1 y L2 y RECALL TIME L1 y L2 selection & options menu detail

3.1.5.2. Net 1 interface.

IP communications interface number 1 over Ethernet port in the rear panel of the unit. There are three working modes available:

- **PROXY SIP:** when the Phoenix STRATOS is used working together with an external SIP server that will provide connection with remote unit through any network, even Internet, without knowing its IP address. Both units (local and remote) must be registered in SIP server.
- **DIRECT SIP:** when you wish to use the SIP communications protocol during the call establishment phase, but are not using any external SIP server. You will need to know the IP address of the unit you're calling to.
- **RTP Point to Point:** when the connection over IP will be an RTP type link with calling of the IP address to IP address type. Obviously there is no advanced signaling protocol in this scenario and you will need to established, set parameters and disconnect communication from both ends; in addition, audio encoding must be the same (and explicitly specified) in both ends of the communication, unless "**SmartRTP**" option is activated (see next section).

Depending on the selected working mode, the rest of the menu options will vary.

3.1.5.2.1. RTP Point to Point.

Submenu for configuring the IP connection when communication is without SIP over audio streaming in RTP.

In the Net X STATUS menu, when "RTP Point to Point" is selected in MODE field, the following options are shown (**NOTE:** The fields that appear in **italics>** cannot be modified, and they are shown as additional information with no editing option available here):



Net1 menu for RTP Point to Point detail

- **ENABLE TRANSMISSION:** In position ON, the transmission of audio is enabled.



ENABLE TRANSMISSION menu detail

- **LOCAL MEDIA PORT:** value of the IP port selected for transmitting audio at origin over IP. Minimum value 1,024. Maximum value 65,534. For instance 1026.



LOCAL MEDIA PORT menu detail

- **DEST MEDIA PORT:** value of the IP port selected for transmitting audio at destination over IP. Minimum value 1,024. Maximum value 65,534. For instance 1026.
NOTE: These values must be crossed (although the four ones can be the same) in both encoding ends in order to be able to establish the bidirectional communication.



DEST MEDIA PORT menu detail

- **SmartRTP MODE (ON or OFF):** the “*SmartRTP*” functionality from **AEQ** greatly simplifies the task of making an RTP call. When operating Phoenix family of AudioCodecs with this mode activated, it is only required that the “caller” launches the call. The remote equipment will automatically answer and send its audio stream to the callers IP and port. Further, the AudioCodec will also detect and automatically select the encoding algorithm that the calling unit is using to initiate the communication. The call does not need to be manually accepted and the hang-up event (from any end) will also be signaled to either unit.



SmartRTP MODE menu detail

- **SYMMETRIC RTP (ON or OFF):** this option forces the local unit to send audio to the same IP and port from which it is receiving audio (it will ignore the IP specified when you call and the port configured in DEST MEDIA PORT). This option will allow you to connect to an equipment with unknown IP and/or port (because it’s behind a router with NAT, for instance). Obviously, the other equipment must know our IP and audio port.



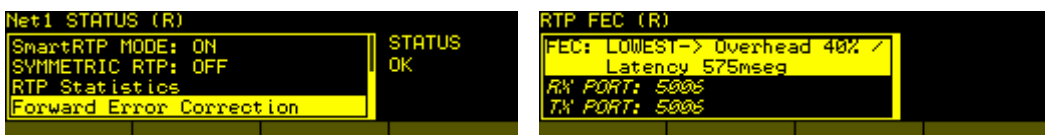
SYMMETRIC RTP menu detail

- **RTP Statistics:** this option provides information regarding RTP communication status when it is established (buffer and jitter size, number of sent/received packets, number of lost packets, number of errors...).



RTP Statistics menu detail

- **Forward Error Correction:** This option enables the different modes of error correction (FEC).



Forward Error Correction menu detail.

For further information on the protocols and technical details involved in communication, we recommend consulting **ANNEXE C** of this manual. Please notice that RTP Point to Point is a complex configuration mode, suitable for permanent connections that some equipment may not support. Specifically, it can’t be used to connect to Phoenix Mobile.

3.1.5.2.2. DIRECT SIP.

Submenu for configuring IP communication when you have a connection with SIP protocol in the signaling phase prior to connection but without the presence of an external SIP server. SIP signaling will take care, once the remote unit has been identified by its IP address and has been reached, of the negotiation of encoding profiles and the communication establishment/cut from any of both ends.



Net 1 menu detail for the DIRECT SIP option

In the Net X STATUS menu, when “DIRECT SIP” is selected in MODE field, the following options are shown (**NOTE:** The fields that appear in **italics** cannot be modified, and they are shown as additional information with no editing option available here):

- **URI:** SIP identifier of the equipment in question. This identifier is “<unit_name>@<unit_IP_address>” type (for instance, “phnstratos_xxx@192.168.1.84”).
When the correspondent **SIP port** is not the 5060 (SIP Standard port) the identifier must include the used port. For instance: “phnstratos_xxx@172.25.32.11:5061”.
This field can be modified by pressing **ADVANCED** in this screen and accessing to USER submenu.



URI options detail

- **RTP Statistics:** this option provides information regarding RTP communication status when it is established (buffer and jitter size, number of sent/received packets, number of lost packets, number of errors...).

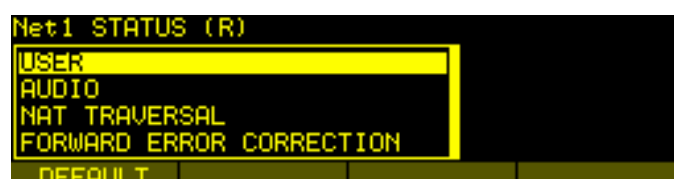


RTP Statistics menu detail

For further information on the protocols and technical details involved in communication, we recommend consulting **ANNEXE C** of this manual and the application notes associated to this working scenario.

ADVANCED menu for DIRECT SIP.

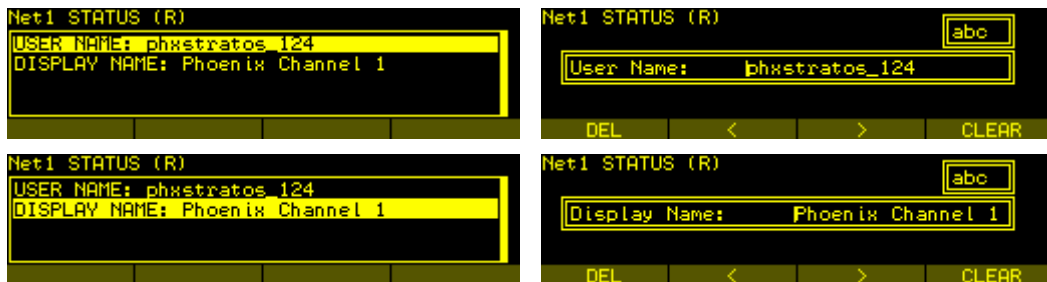
The ADVANCED option associated with the fourth contextual button in the lower part of the screen allows you to access the advanced configuration of the IP options of the Phoenix STRATOS unit, where you can configure the following parameters for DIRECT SIP working mode:



ADVANCED menu for direct SIP detail

- **USER:** enables you to edit the name of the unit and how it will be reflected in the diverse internal menus of the unit.

There are two fields: “USER NAME” and “DISPLAY NAME”. For a start we recommend you not to change the configured default User Name, “phxstratos_001” for instance.

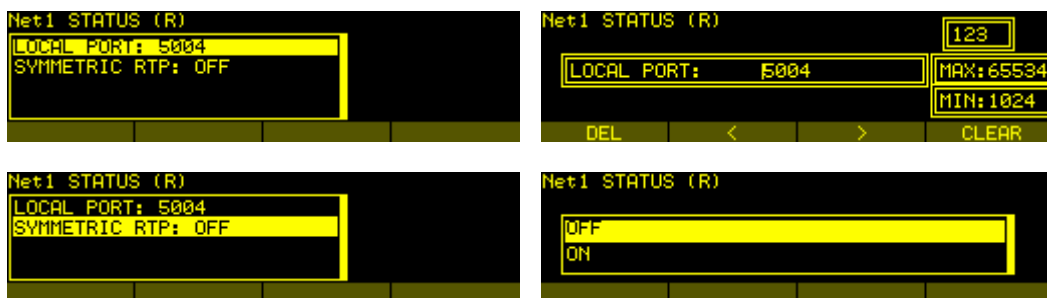


USER menu options

- **AUDIO:** The “LOCAL PORT” option allows you to configure the value of the IP port selected to transmit audio at origin over IP. Minimum value 1,024. Maximum value 65,534. Usually recommended value 5004.

NOTE: Do not repeat values that were previously associated with SIP signaling.

The “SYMMETRICAL RTP” (ON or OFF) option allows you to force the local unit to send audio to the same IP and port from which it is receiving audio. This option will allow you to connect to an equipment with unknown IP and/or port (because it’s behind a router with NAT, for instance).



AUDIO menu detail

- **NAT TRAVERSAL:** see section “3.1.5.2.4 NAT TRAVERSAL submenu”.

IMPORTANT NOTE: Due to the complexity inherent in most of the parameters involved in this ADVANCED menu and the importance that any modification has in the final operation of the unit, we recommend that only highly qualified personnel in possession of all the technical documentation and manuals work on this advanced configuration menu.

3.1.5.2.3. PROXY SIP.

Submenu for configuring IP communication when you have a connection with SIP protocol in the signaling phase prior to connection and with the presence of an external SIP server, which function is to maintain a data base of users codecs registered with their connection parameters (IP address, audio ports, encoding algorithms...) in order to make easier the connection between codecs previously registered, even when they work in different networks.



Net 1 for PROXY SIP menu detail

In the Net 1 STATUS menu, when “PROXY SIP” is selected in MODE field, the following options are shown (**NOTE:** The fields that appear in *italics>* cannot be modified, and they are shown as additional information with no editing option available here):

- **URI:** SIP identifier of the equipment in question. This identifier is “<unit_name>” or “<unit_name>@<SIP_server_name/IP>[:<Port>]” type. For instance:
 - o “phoenixMaster@sip.aeq.es”
 - o “phxSTRATOS”
 - o “phxst_035@sip.aeq.es:5062”
 - o “phxst_035@88.25.3.14” (where 88.25.3.14 is the SIP server IP, not the one of the unit you’re calling to)

When the SIP port of SIP server is not the 5060 (SIP Standard port) the identifier must include the used port. For instance: URI: “phxst_123@sip.aeq.es:5061”.

This field can be modified by pressing **ADVANCED** in this screen and accessing to USER submenu.

For further information on the protocols and technical details involved in communication, we recommend consulting **ANNEXE C** of this manual and the application notes associated to this working scenario.

ADVANCED menu for PROXY SIP.

The ADVANCED option associated with the fourth contextual button in the lower part of the screen allows you to access the advanced configuration of the IP options of the Phoenix STRATOS unit, where you can configure the following parameters for PROXY SIP working mode:



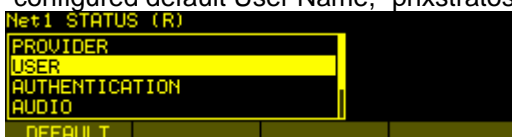
ADVANCED menu options

- **PROVIDER:** enables you to select the external SIP server with which the unit will work from a previously stored list. See section “3.1.9. SIP PROVIDERS menu” in this manual. By pressing EDIT button you can access to the PROVIDERS edition screen described in that section; you can also access to this screen through SYSTEM / 9 – SIP PROVIDERS. By default, AEQ server will be selected.



PROVIDER menu options

- **USER:** enables you to edit the name of the unit and how it will be reflected in the diverse internal menus of the unit. For a start we recommend you not to change the configured default User Name, “phxstratos_124” for instance.



USER menu detail

- **AUTHENTICATION:** enables you to edit the password and security information for the user profile associated with the Phoenix STRATOS unit in the previously selected SIP server.

By default, the data configured in this field in order to use AEQ server are the following:

USER: the “User Name” configured in Factory, “phstr_1” for instance.

PASSWORD: the one associated to that user.

REALM or DOMAIN: sip.aeq.es



AUTHENTICATION menu options

- **AUDIO:** The “LOCAL PORT” option allows you to configure the value of the IP port selected to transmit audio at origin over IP. Minimum value 1,024. Maximum value 65,534. Usually recommended value 5004.



AUDIO – LOCAL PORT menu detail

The “SYMMETRICAL RTP” (ON or OFF) option allows you to force the local unit to send audio to the same IP and port from which it is receiving audio. This option will allow you to connect to an equipment with unknown IP and/or port (because it’s behind a router with NAT, for instance).



AUDIO – SYMMETRICAL RTP menu detail

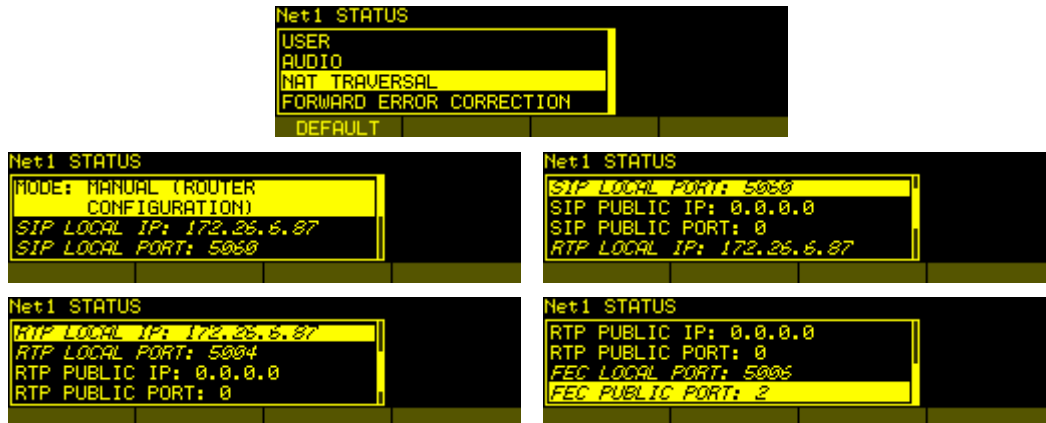
- **NAT TRAVERSAL:** see section “3.1.5.2.4 NAT TRAVERSAL submenu”.

Due to the technical complexity inherent in most of the parameters involved in this ADVANCED menu and the importance that any modification has in the final operation of the unit, we recommend that only highly qualified personnel in possession of all the technical documentation and manuals work on this advanced configuration menu.

3.1.5.2.4. NAT TRAVERSAL submenu.

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

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



NAT TRAVERSAL options menu detail

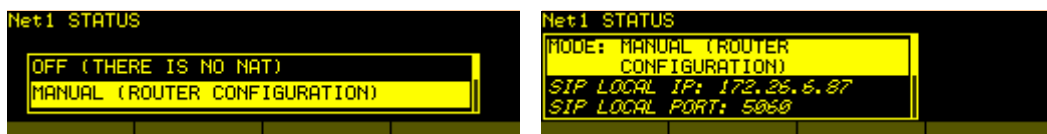
Detailed descriptions are given below of the working modes supported by Phoenix STRATOS:

- **OFF (LOCAL NETWORK):** The unit uses no mechanism to traverse devices with NAT. This mode will be used only to operate in the local network (all of the SIP participants are in the same local network, including the Proxy SIP).



Mode OFF selection detail

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



Mode MANUAL selection detail

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

AEQ PHOENIX AUDIOCODECS. APPLICATION NOTE 0 - C (Connecting AEQ Phoenix STRATOS units through Internet. Complex scenario configuration. Through LOCAL network(s), DHCP not used, manual NAT. Making use of AEQ Proxy SIP).

The eight configuration parameters available on screen for this mode are:

1. **SIP LOCAL IP:** read-only parameter that tells you the IP of the IP interface of the unit as regards SIP, so that the latter can, in turn, convey this to the router or firewall administrator when it is configured. For instance 172.26.5.59. It can be set in order to adapt it to network necessities in menu / SYSTEM / 8.SETTINGS / ETHERNET SETTINGS / ETHERNET MODULE

```

Net1 STATUS
MODE: MANUAL (ROUTER
CONFIGURATION)
SIP LOCAL IP: 172.26.6.87
SIP LOCAL PORT: 5060
    
```

SIP LOCAL IP detail

2. **SIP LOCAL PORT:** read-only parameter that tells you the port of the IP interface of the unit used for SIP signaling, so that the latter can, in turn, convey this to the router or firewall administrator when it is configured. Before checking the value of this parameter you should have configured previously whether you want to work with Proxy or not and restart the unit.

```

Net1 STATUS
MODE: MANUAL (ROUTER
CONFIGURATION)
SIP LOCAL IP: 172.26.6.87
SIP LOCAL PORT: 5060
    
```

SIP LOCAL PORT detail

3. **SIP PUBLIC IP:** parameter that will tell the unit which public IP will correspond to it, so that it can include the said IP in its SIP messages. The router or firewall administrator must tell you the value of this parameter. For instance 212.170.180.177.

<pre> Net1 STATUS SIP LOCAL IP: 172.26.6.87 SIP LOCAL PORT: 5060 SIP PUBLIC IP: 0.0.0.0 SIP PUBLIC PORT: 0 </pre>	<pre> Net1 STATUS SIP IP: 0.0.0.0 1.2.3 DEL < > CLEAR </pre>
---	--

SIP PUBLIC IP menu detail

4. **SIP PUBLIC PORT:** parameter that will tell the Phoenix STRATOS which public port it will have corresponding to its local SIP port. The router or firewall administrator must tell you the value of this parameter in order to make the required port-forwarding. For instance: 8001.

<pre> Net1 STATUS SIP LOCAL IP: 172.26.6.87 SIP LOCAL PORT: 5060 SIP PUBLIC IP: 0.0.0.0 SIP PUBLIC PORT: 0 </pre>	<pre> Net1 STATUS PORT: 0 123 MAX: 65534 MIN: 1024 DEL < > CLEAR </pre>
---	---

SIP PUBLIC PORT menu detail

5. **RTP LOCAL IP:** read-only parameter that tells you the IP of the IP interface of the unit as regards RTP, so that it can, in turn, convey this to the router or firewall administrator when it is configured. You will usually configure the same network interface as for SIP, so it will be the one configured in point number 1: for instance 172.26.5.59.

```

Net1 STATUS
SIP LOCAL PORT: 5060
SIP PUBLIC IP: 0.0.0.0
SIP PUBLIC PORT: 0
RTP LOCAL IP: 172.26.6.87
    
```

RTP LOCAL IP detail

6. **RTP LOCAL PORT:** read-only parameter that tells you the port of the IP interface of the unit as regards RTP, so that the latter can, in turn, convey this to the router or firewall administrator when it is configured. Usually the shown port is 5004.

```

Net1 STATUS
SIP PUBLIC IP: 0.0.0.0
SIP PUBLIC PORT: 0
RTP LOCAL IP: 172.26.6.87
RTP LOCAL PORT: 5004
    
```

RTP LOCAL PORT detail

7. **RTP PUBLIC IP:** parameter that will tell the unit which public IP will correspond to the RTP of its IP interface, so that it can send the said IP in its SIP messages. The router or firewall administrator must tell you the value of this parameter. Usually the administrator will take out SIP traffic and RTP using the same public IP configure in point number 3. For instance 212.170.180.177.



RTP PUBLIC IP menu detail

8. **RTP PUBLIC PORT:** parameter that will tell the Phoenix STRATOS which public port will correspond to the RTP of its IP interface, so that it can send the said port in its SIP messages. The router or firewall administrator must tell you the value of this parameter in order to make the required port-forwarding. For instance: 8002.



Detalle menu RTP PUBLIC PORT

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

- **AUTO 1 (LOCAL AUDIO).** This mode will be used primarily when two units that are in the same local network need to communicate with each other, when the Proxy SIP is on the Internet and it's the one provided by AEQ (sip.aeq.es).
- **AUTO 2 (LOCAL AUDIO).** This mode will be used primarily when two units that are in the same local network need to communicate with each other, when a Proxy SIP on the Internet is used, it's not the one provided by AEQ (sip.aeq.es) and only if the AUTO1 mode doesn't work properly.
- **AUTO 3 (AUDIO THRU INTERNET).** This mode will be used mainly when you wish to put two units in communication with each other through the Internet, working with no Proxy (DIRECT SIP mode) or using the Proxy SIP provided by AEQ (sip.aeq.es) (PROXY SIP mode). The two configuration parameters available on screen for this mode are:
 1. **STUN SERVER:** parameter that tells the unit the STUN server that will be used. On the Internet there is multitude of public STUN servers. By default, the IP address of stun.sipgate.net server is configured: 217.10.68.152.
 2. **STUN PORT:** parameter that tells the unit the STUN server port assigned by the administrator. By default: 3478.

NOTE: in this mode the Phoenix STRATOS behaves in the exact same way as the Phoenix Mobile unit when it is using a STUN server.
- **AUTO 4 (AUDIO THRU INTERNET).** This mode is equivalent to AUTO3 but it will be used the SIP servidor is not the one provided by AEQ and there are problems with AUTO3 mode. The configuration parameters are the same as for AUTO3 (STUN server specification).

Due to the technical complexity inherent in most of the parameters involved in this NAT TRAVERSAL menu and the importance that any modification has in the final operation of the unit, we recommend that only highly qualified personnel in possession of all the technical documentation and manuals work on this NAT configuration menu. For additional information, see **ANNEXE C5**.

3.1.5.3. Net 2 interface.

IP communications interface number 2 over Ethernet port in the back panel of the unit. The options in this menu are exactly the same as those described for the Net1 interface.

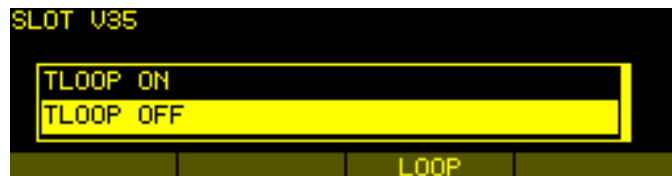
3.1.5.4. V35 interface.

For the default V35 interface, select this option to access the menu where you can modify the working mode by choosing PERMANENT or DCD CONTROLLED. For further information on the V.35 interface, see section 2.2.5 of this manual.



V.35 interface menu detail

The option displayed in the lower part of the screen (LOOP) enables you to create a loop in the V.35 communication interface, with applications in the maintenance of the codec unit or tests conducted before installing and evaluating the equipment.



V.35 LOOP options menu in V.35

3.1.6. CHANNELS menu.

Configuration menu for the two communication channels available in the PHOENIX STRATOS, indicating the communication interface used and the encoding algorithm employed, among many other parameters. You can access the CHANNELS menu from the SYSTEM MENU, in the sixth option, by pressing the rotary encoder. You can also access to channels configuration from the MAIN STATUS screen by pressing the encoder when placed over Chn1 or Chn2.



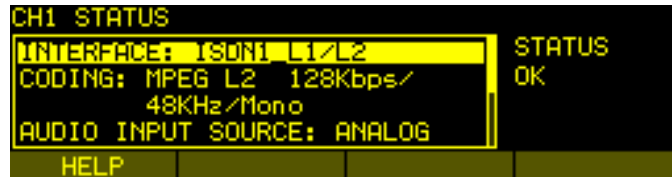
CHANNELS menu detail

The PHOENIX STRATOS unit analyzed in the previous image is prepared to transmit communication channel number 1 (Chn1), over the ISDN1_L1/L2 communication interface, with the MPEG L2 128Kbps encoding algorithm at a sampling rate of 48KHz Mono, while communication channel 2 has Net2 logic IP interface associated with it and it's connected to IP address 172.26.33.27 with G722 mode.

To shift between the two entries in this menu, for channels 1 and 2, turn the rotary encoder. Pressing the rotary encoder accesses the configuration of the currently selected channel.

To quit this menu and return to the immediately previous one, simply press the ESC button.

According to the type of communication interface selected (IP, ISDN or V35), the number of options available in this menu may vary, although all of them are detailed below:



CHANNEL 1 configuration detail

- **INTERFACE:** lets you select the communications interface from among the ones available by default in the unit: Net 1 (IP), Net 2 (IP), V35, ISDN1_L1, ISDN1_L2. Una de las posiciones del selector está inactiva y aparecerá la indicación “NO INTERFACE”.



Selection options associated with the communication interfaces

In the case of ISDN interfaces, the two B channels that comprise them are displayed separately as L1 and L2 for communications at 64Kbps or ISDN1_L1/L2 for communications at 128Kbps (as shown in the previous image).

- **CODING / CODING (List):** enables you to select the encoding algorithm associated with this communication channel. Depending on the communication interface selected, certain encoding modes may not be available.



List of algorithms

In the case of a SIP type communication over Net 1 or Net 2 interface, the isolated encoding concept is replaced by the encoding profile concept, that is, an encoding list to be negotiated between the units at both ends during the SIP phase of signaling, so instead of specific encoding modes you choose modes groups (profiles) that are created separately. For further information, consult section “3.1.10. SIP CODECS PROFILES menu” of this manual.

- **AUDIO IN SOURCE:** allows you to select the source of audio of the communications channel: ANALOG input(s) or DIGITAL input.



Audio input configuration options

- **BACK UP:** this submenu enables you to configure the switching from the currently selected communication interface toward a redundant communication interface in case of connection failure. For a detailed description of this submenu, see section 3.1.6.1 of this manual.



BACKUP option configuration

- **AUTORECALL MODE:** this submenu allows you to specify the way the selected communications channel will act when the unit starts up: no action or establish back the last active call when the unit was turned off (due to a power supply failure, for instance), in several way. This function, combined with BACK UP function, provides ways to ensure a high disponibility link, solid to face to any kind of trouble that may appear. Please consult section 3.1.6.2 of this manual for a detailed description.
- **RX BUFFER MODE (only for IP interface):** configuration of the reception buffer as adaptive or fixed; in the former case its size will be variable according to the transmission network conditions, or fixed at value X according to manual configuration. This option will appear only when IP interface is selected, because V35 and ISDN modes are synchronous and there is no need to delay the audio because reception is guaranteed with no network delay when there is no communication failure.



RX BUFFER MODE menu detail

- **RX BUFFER (MAX) (only for IP interface):** maximum size value of the reception buffer. In the case of an adaptive buffer, from this value upward the Phoenix STRATOS would begin to adjust itself on the basis of the transmission network status. If BUFFER FIX (fixed) is defined, this maximum value will be the same as the minimum value, since the size will not change during the entire duration of the connection. Value in milliseconds. The higher the buffer, the lower the probability of packets loss, but in addition the delay will be higher, specially when the selected mode is FIX.



Configuration of the maximum reception buffer size

- **AUTOANSWER (IP interface) / OFF HOOK (ISDN interface):** allows the automatic answer mode to be activated (ON for IP, AUTOMATIC for ISDN). This option will not appear when V35 interface is selected.
- **AUTOANSWER URI (IP interface) / OFF HOOK NUMBER (ISDN interface):** enables the automatic answer mode to be associated with an IP identifier (URI) of the calling equipment (or with a number or pair of numbers -128Kbps- in case of an ISDN connection). When calls come in from other URIs (or ISDN numbers), the unit reports their reception and behaves as it would be in manual answering mode, never rejecting any incoming call.
NOTE: if you introduce an IP identifier, it should be the <unit_name>@<SIP_server_domain> type (for instance, "phstr_xxx@sip.aeq.es"), it's NOT enough to use just the <unit_name> as identifier.
 This option will not appear when V35 interface is selected.

- **RTP INACTIVITY MODE (only for IP interface):** allows you to enable the detection of audio signal absence in audio streaming over IP. This option will appear only when IP interface is selected. When activated (and it is configured this way by default), the unit will terminate the call when no RTP traffic is received after a certain time (specified in the next option). This way you can avoid that the unit gets blocked when a call signaling has been lost, for instance.
- **RTP INACTIVITY TIME (sec) (only for IP interface):** configures the minimum waiting time that the Phoenix STRATOS must wait before considering that the absence of audio over RTP is final and irreversible and terminates the call (only when RTP INACTIVITY MODE is ON). Values in seconds. The equipment is supplied with a default configuration of 15 seconds. This option will appear only when IP interface is selected.

```

CH1 STATUS
-----
AUTOANSWER: ON
AUTOANSWER URI:
RTP INACTIVITY MODE: OFF
RTP INACTIVITY TIME(sec): 30
-----
HELP
  
```

Exclusive configuration options for an IP interface

3.1.6.1. BACK UP submenu.

The BACK UP submenu offers the following options:

- **ENABLE:** ON or OFF can be selected. Selecting OFF indicates that you do not wish to use this redundancy option. Selecting ON gives meaning to the rest of the parameters that are described below (the ON option is available only when a backup interface is previously selected).
- **INTERFACE*:** lets you select a free interface as a backup toward which to switch the communication if there is a connection failure. When all the communication interfaces are busy, the “NO INTERFACE FREE” message will be displayed.
- **CODEC / CODING (List):** allows you to configure the encoding algorithm (or algorithm list for IP interface configured in “PROXY SIP” or “DIRECT SIP” mode) to be used over the selected backup interface. Obviously you can select a different algorithm in this backup interface from the one selected in the main interface.
- **CALL TIME OUT:** defines the minimum time interval for the connection to be considered down and the communication interface switching to be performed.
- **RETRY ATTEMPTS:** lets you select the number of retries to be made before considering the communication down.
- **SYNC TIME OUT:** defines the minimum time interval for the connection to be considered down if synchronization is not performed, and the communication interface switching to be performed.
- **CONTACT / URI** (for IP interface configured in “PROXY SIP” or “DIRECT SIP” mode) / **DEST IP** (for IP interface configured in “RTP Point to Point” mode): defines the destination toward which the backup connection is to be launched. Depending on the communication interface you have selected as a backup, this destination will be defined by an ISDN subscriber number, a SIP identifier (URI) or an IP address. This option will not appear when V35 interface is selected.

***NOTE:** when you only want that the call using the main interface will be retried in case of a connection failure, you just have to select a backup interface that you’re not using (V35, for instance) and configure ENABLE as ON. This way, when communication is lost, the unit will retry to connect (RETRY_ATTEMPTS times) and, when it’s not successful, will change to V35 interface. Since there is nothing connected to this interface, after the same number of retries, it will change back to main interface and it will go on this way until the line is available again.

BACK UP function can be used together with AUTORECALL option (see next section) in order to guarantee the link stability against different failure situations, such as line falls or power supply failures.

3.1.6.2. AUTORECALL MODE submenu.

AUTORECALL MODE function defines the way a communication channel will act when the unit starts up. There are 4 different modes and can be independently configured for both communication channels:

- **DISABLED:** the channel establishes no automatic communication at all (by default).
- **ALWAYS CALL TO:** when the unit starts up, the channel launches a call to a previously specified number/URI. The number/URI that will be used is the one that appears when you press the CALL button of the selected channel. Therefore, in order to configure this option, press CALL just like if you were about to make a call to the desired equipment, configure the desired number/URI, press ESC and enter the channel menu in order to activate this mode.
- **REDIAL INTERRUPTED CALL:** when the unit starts up, the channel launches an automatic call as long as the communication with that number were established when the unit was turned off (due to a power supply failure, for instance).
- **REDIAL INTERRUPTED OUTGOING CALL:** when the unit starts up, the channel launches an automatic call as long as the communication with that number were established when the unit was turned off (due to a power supply failure, for instance) and that call were also made by the local unit.

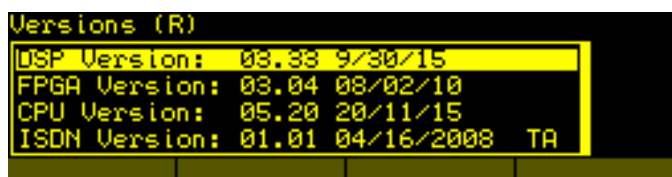


NOTE: When the channel is working in IP - RTP mode, the call will be reestablished only when the remote unit is still connected, but it could be disconnected when RTP INACTIVITY MODE is activated and enough time has passed by. In this case, we recommend you to activate additionally the BACK UP function in order to guarantee a failureproof link.

3.1.7. MAINTENANCE menu.

Internal maintenance menu for the codec. You can access the MAINTENANCE menu from the SYSTEM MENU, in the seventh option, by pressing the rotary encoder. It is internally divided into a series of submenus:

- **FW VERSION:** enables you to consult the firmware versions of the various PHOENIX STRATOS basic modules (DSP, FPGA and CPU), as well as the ISDN module. More information in section 1.4.



- **REMOTE CONTROL SETTINGS:** allows you to configure the unit to accept incoming remote control connections. Make sure that there is connection with the PC where "AEQ ControlPHOENIX" application is installed, the display headline shows an "(R)" to indicate it (independently of the menu we have selected).



- **CODEC LOOP TEST CH1:** allows you to set up an encoded audio loop from the inputs to the outputs of channel 1 of the unit (according to configuration in AUDIO SETTINGS, ANALOG AUDIO SETTINGS and CHANNELS), to verify that it is operating correctly, with applications in the maintenance of the unit or tests conducted before installing and evaluating it.
- **CODEC LOOP TEST CH2:** allows you to set up an encoded audio loop from the inputs to the outputs of channel 2 of the unit (according to configuration in AUDIO SETTINGS, ANALOG AUDIO SETTINGS and CHANNELS), to verify that it is operating correctly, with applications in the maintenance of the unit or tests conducted before installing and evaluating it.
- **HW VERSION:** version identifier of the hardware platform on which the Phoenix STRATOS unit in question is working. At the time this manual was being prepared, the latest available version is V11.
- **BUZZER TEST:** pressing the rotary encoder activates the ringing signal, to verify that it is functioning properly.
- **LINEAR AUDIO LOOP CH1:** allows you to set up a loop from the inputs to the outputs of channel 1 (according to configuration in AUDIO SETTINGS, ANALOG AUDIO SETTINGS and CHANNELS), with applications in the maintenance of the unit or tests conducted before the unit is installed and evaluated. Similar to the CODEC LOOP TEST feature, but with linear audio.
- **LINEAR AUDIO LOOP CH2:** allows you to set up a loop from the inputs to the outputs of channel 2 (according to configuration in AUDIO SETTINGS, ANALOG AUDIO SETTINGS and CHANNELS), with applications in the maintenance of the unit or tests conducted before the unit is installed and evaluated. Similar to the CODEC LOOP TEST feature, but with linear audio.
- **SYSLOG SETTINGS:** allows you to activate and configure the remote logs sending function. You can activate/deactivate it, select the physical interface to send those and the destination IP corresponding to a PC where the client SysLog application should be installed. This option is used for high level debugging and technical support. By default, this option is deactivated and it should not be activated unless AEQ's Technical Service requests for it.
- **SAVE SETTINGS:** allows you to save the system configuration.
- **RESTORE SETTINGS:** allows you to load the previously saved configuration.
- **DEFAULTS SETTINGS:** allows you to load the default system configuration.

3.1.8. SETTINGS menu.

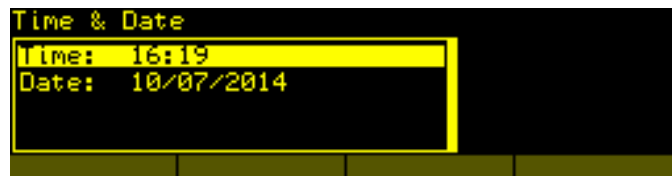
Menu for the local configuration of the unit, including Ethernet configuration. You can access the SETTINGS menu from the SYSTEM MENU, in the eighth option, by pressing the rotary encoder. This menu enables you to configure the following parameters.

3.1.8.1. Audible call notification.

The BUZZER submenu lets you activate/deactivate the incoming call notification buzzer.

3.1.8.2. Date and time.

The TIME & DATE submenu allows you to configure the time and date of the unit's local system.



The procedure to configure the time in the Phoenix STRATOS codec is quite simple. Starting from the default start screen, you can use the rotary encoder to select the modification of the information associated with the time or the date. In both cases the lower line on the screen represents actions associated with the contextual buttons.

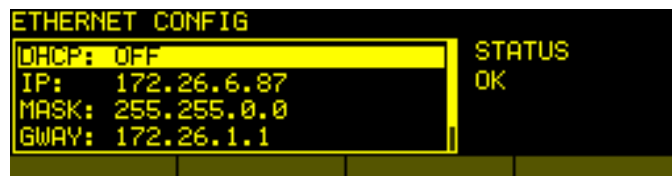
- **DEL:** deletes the character placed on the left side of the cursor (except when the cursor is placed at the beginning: in that case it deletes the character placed on the right side of the cursor).
- **<:** lets you move the writing cursor to the left.
- **>:** lets you move the writing cursor to the right.
- **OK:** validates the entered value.

To create the date or time, simply dial the desired digits using the alphanumeric keypad.

For the date, pressing the rotary encoder enables you to modify, in a single round, the data relative to the day, month and year, moving consecutively from one screen to the next. From the default start screen, pressing the ESC key enables you to return to the previous SETTINGS menu.

3.1.8.3. Ethernet port configuration.

The ETHERNET SETTINGS submenu allows you to configure the IP parameters of the Ethernet port in the unit.



Detail of the IP configuration of an Ethernet port

The interface offer the configuration options:

- **DHCP:** enables the activation or deactivation of the automatic configuration of IP addresses, masks and gateways. There must be a DHCP server in the network the unit is connected to in order to make this option work. When DHCP=ON the following parameters will be filled automatically; when DHCP=OFF you will be able to change them manually.
- **IP:** valid IP address associated with the interface.
- **MASK:** valid subnet mask associated with the interface.
- **GWAY:** an abbreviation of GATEWAY, valid gateway or network gateway address associated with the interface.
- **DNS:** IP address of the external addresses resolution server, valid in the geographic zone where codec is placed, or internal server (inside the local network) authorized to translate alphanumeric URL identifiers into IP addresses.

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

3.1.8.4. GPIOs configuration (GPIO Settings).

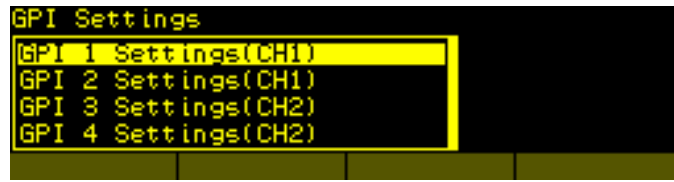
The GPIOs SETTINGS submenu allows you to read and configure the status of the general purpose inputs and outputs.



GPIO configuration submenu

There are 4 GPIs (General Purpose Input) and 4 GPOs (General Purpose Output). Each input and output is associated with a communication channel. In this submenu, you can define the outputs performance according to certain events in the associated channel and that channel performance when there are changes in the corresponding input.

- **GPIs Settings:** This submenu allows you to configure the GPI inputs functionality. When you enter this menu, another submenu will appear allowing you to select the inputs GPI 1 and GPI 2 associated to channel 1, GPI 3 and GPI 4 associated to channel 2.



The four menus are identical and consist of the following options:



- **GPI Function:** there are several possible reactions of the unit when there's a change in the associated input:
 - CALL Channel x: it will work the same way the "CALL" button of the associated channel does.
 - GPIx_value: it shows the current GPI value.



GPIs function configuration submenu

- **GPI Mode:** indicates the way the unit will react to the associated GPI:
 - Pulse mode GPIx: the function selected in GPI Function will be activated when a pulse is detected.
 - Toggle mode GPIx: the function selected in GPI Function will be activated when the input level changes from "0" to "1" logical value.



GPIs operation mode configuration submenu

- **GPOs Settings:** This option allows you to configure the GPO outputs functionality. When you enter this menu, another submenu will appear allowing you to select the 4 outputs, two associated to channel 1 and two to channel 2. The four menus, consist of the following options:

- Incoming call in Channel x: the corresponding GPO will be activated when an incoming call is detected.
- Force to 0
- Force to 1

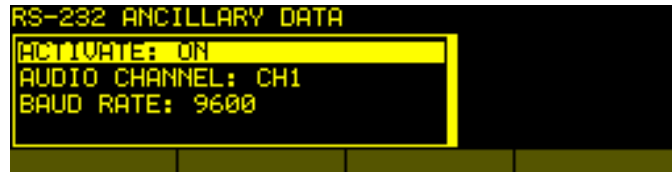


GPOs operation mode configuration submenu

3.1.8.5. RS-232 Ancillary Data.

The PHOENIX STRATOS allows you to perform a bidirectional data transmission between units by means of COM rear connector. This option may be useful in order to send files from one location to another, or also to control remotely other units with RS232 serial interface. Data are embedded in audio streams. At the moment, this option is available for ISDN communications (at 64 and 128Kbps) with MPEG-1 y 2 LII encoding modes (bit rates: 16, 24, 32 and 48 KHz).

The “RS-232 Ancillary Data” menu allows you to configure wheter the ancillary data transmission will be used or not (ACTIVATE ON/OFF), as well as the associated audio channel (AUDIO CHANNEL: CH1 or CH2). Transmission speed is also shown as additional information (it can't be modified), fixed at 9600 bps (BAUD RATE = 9600).



Ancillary Data menu detail

This information transmission does not cause a noticeable loss of audio quality, specially when using the 128Kbps modes.

3.1.8.6. Remote configuration.

The WEB SERVER USERS submenu enables you to configure the name and password associated with the remote control and configuration WebBrowser user. To obtain more information, we recommend consulting section “7. CONTROL TERMINAL OVER WEB BROWSER” in this manual.



WEB SERVER USERS menu detail

3.1.9. SIP PROVIDERS menu.

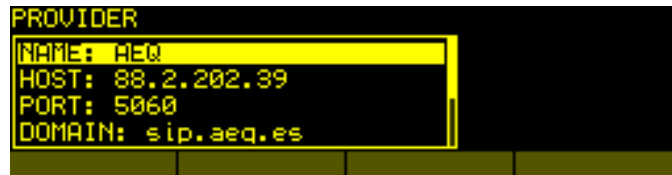
Configuration menu for the parameters associated with operation over an external SIP server. Accessible from the SYSTEM MENU, in the ninth option, by pressing the rotary encoder.

In the SIP PROVIDERS menu you can store up to eight different configurations to work with as many as eight different external SIP servers. As indicated in chapter one of this manual, AEQ offers, at no additional cost, the services of its **2 dedicated SIP servers**, and therefore the first 2 entries in the SIP PROVIDERS menu include the relevant values, leaving the rest of the entries at your disposal for future configurations. For further information, see **ANNEXE C4.3**.



SIP PROVIDERS menu detail

- **NAME:** identifier of the SIP server on the list of suppliers. For instance, AEQ.
- **HOST:** SIP server host name. For instance, "sip.aeq.es". You can also configure directly its IP address.
- **PORT:** IP port where the configuration is performed. Typically 5060/5061.
- **DOMAIN:** IP domain under which the Phoenix STRATOS unit will work when it connects to the external SIP server. For instance, "sip.aeq.es".
- **REGISTER:** enables the option of registering the Phoenix STRATOS unit in the SIP server. Only highly specific configurations and very particular cases that fall outside the objective of this manual may require the disabling of this option.
- **EXPIRES:** length in seconds of the register of the unit in the remote SIP server. Once this period has ended, the unit will renew its register. We recommend using low values in this field. For instance, 60.



SIP PROVIDERS configuration menu options

3.1.10. SIP CODECS PROFILES menu.

You can access this menu from the SYSTEM MENU, in the tenth and last option, by pressing the rotary encoder. Menu for configuring the parameters associated with encoding to be used in an audio connection over IP networks based on SIP protocol (Proxy SIP and Direct SIP modes). This option simplifies the selection of the algorithm to be used in a communication, because most of the codecs have several tens of encoding algorithms in order to have the higher compatibility with other equipments.

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



SIP CODECS PROFILES menu detail

Each one of the stored entries includes an alphanumeric identifier and a list of algorithms to use, organized in order of preference. There several preset profiles in the unit, grouped by delay, quality... Obviously, profiles can be added, modified or deleted.

You can shift among the various list entries by turning the rotary encoder. You can okay an option by pressing on the rotary encoder.

There are four options displayed in the lower part of the screen associated with the four contextual buttons located beneath the OLED display.

- **INSERT:** allows you to create a new encoding profile. The system requests a name and algorithm list to be used in order of preference.
- **DEL:** enables you to eliminate a previously created encoding list. The system will request confirmation of this action.
- **VIEW:** lets you see, without modifying them, the contents of an encoding list.

- **EDIT:** allows you to edit the contents of an encoding list by modifying the list identifier name (EDIT), modifying the algorithms that comprise it (by adding an algorithm with INSERT or eliminating one with DEL) and by changing their order of preference (by pressing MOVE, then selecting a line and moving it by means of the rotatory encoder and finally pressing ENTER to place it in its new position).



List of algorithms of a SIP CODEC PROFILE and editing options

To quit this menu and return to the immediately previous SYSTEM MENU, simply press the ESC button.

3.2. HPHONES menu.

From this option of MAIN STATUS menu you can assign any of the signals available in the PHOENIX STRATOS to the headphones output available in the front panel. This headphones output for control monitoring enables you to associate different signals in the left and right channels.

The HPHONES menu screen, in addition to allowing you to configure this monitoring output, displays the current configuration in the right-hand portion of the screen before making any change (in the example, the left headphone monitorizes the left input to channel 1 and the right one monitorizes the right output of channel 2).



HPHONES menu detail

By pressing the rotatory encoder over “L CHANNEL”, you can select the signal that you will listen on HP_L (left headphone):

- **IN_CH1_L:** Left audio input to communication channel 1.
- **OUT_CH1_L:** Left audio output of communication channel 1.
- **IN_CH2_L:** Left audio input to communication channel 2.
- **OUT_CH2_L:** Left audio output of communication channel 2.
- **IN_CH1_R:** Right audio input to communication channel 1.
- **OUT_CH1_R:** Right audio output of communication channel 1.
- **IN_CH2_R:** Right audio input to communication channel 2.
- **OUT_CH2_R:** Right audio output of communication channel 2.

You can shift among the various options by turning the rotary encoder. You can okay an option by pressing on the rotary encoder.

By pressing the rotatory encoder over “R CHANNEL”, you can select the signal that you will listen on HP_R (right headphone) from an identical list.

To quit this menu and return to the immediately previous MAIN STATUS menu, simply press the ESC button.

Headphones volume can be modified by means of the potentiometer placed besides the corresponding connector in the front panel.

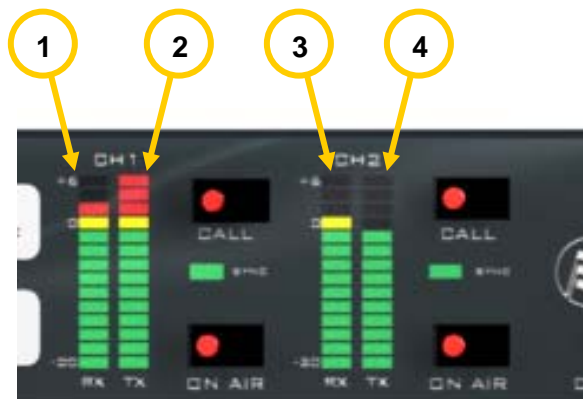
3.3. VUMETERS menu.

From this option of MAIN STATUS menu you can configure the working mode of the two groups of VU meters available in the front panel of the unit.

There are three working modes available in the Phoenix STRATOS:

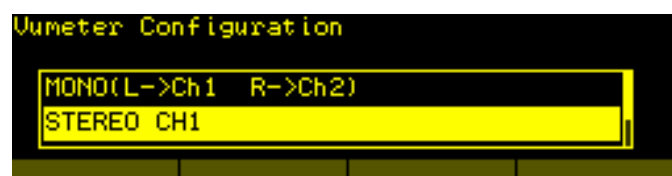
- **MONO (L->CH1 R->Ch2)**, in which the VU meters would look as follows (from left to right):
 - VU1: Output CH1 (received signal)
 - VU2: Input CH1 (signal to be transmitted)
 - VU3: Output CH2 (received signal)
 - VU4: Input CH2 (signal to be transmitted)
- **STEREO CH1**, in which the VU meters would look as follows (from left to right):
 - VU1: Input L CH1 (signal to be transmitted)
 - VU2: Input R CH1 (signal to be transmitted)
 - VU3: Output L CH1 (received signal)
 - VU4: Output R CH1 (received signal)
- **STEREO CH2**, in which the VU meters would look as follows (from left to right):
 - VU1: Input L CH2 (signal to be transmitted)
 - VU2: Input R CH2 (signal to be transmitted)
 - VU3: Output L CH2 (received signal)
 - VU4: Output R CH2 (received signal)

NOTE: MONO option is the only one where signals correspond to vumeters serigraphed lettering.



Identification of the VU meters in the Phoenix STRATOS front panel

The last two working modes are obviously optimized for stereo connections, while the first allows you to monitor two full-duplex mono signals simultaneously. You can change configurations by pressing on the rotary encoder.



VUMETERS menu options

The VIEW option available in the second contextual button offers you a detailed view of the configuration you have selected at any given moment.

3.4. CALL menu.

Menu for configuring the connection to be launched. You can access this menu by pressing either of the two CALL buttons located on the PHOENIX STRATOS front panel.

The left-hand CALL button is associated with communication channel 1, while the right-hand CALL button is linked with communication channel 2.



CALL menu detail (ISDN 128Kbps)

“CH1 CALL MENU” and “CH2 CALL MENU” menus has seven or eight configurable fields in case of an IP communication (the first four initially visible and the following three or four available by turning the encoder) and only four (the first four) in case of a V35 or ISDN communication, with different content depending on the communication type (SIP, IP RTP or ISDN):

- **URI / DEST IP /TLF (_L1/_L2):**
 - **URI** (IP interface in “PROXY SIP” or “DIRECT SIP” mode): IP identifier (URI).
 - **DEST IP** (IP interface in “RTP Point to Point” mode): IP address format x.x.x.x.
 - **TLF:** destination ISDN number (TLF_L1 and TLF_L2 in case of a 128Kbps ISDN communication: both numbers must be filled in).

NOTE: in the case of an IP identifier, the usual format is:

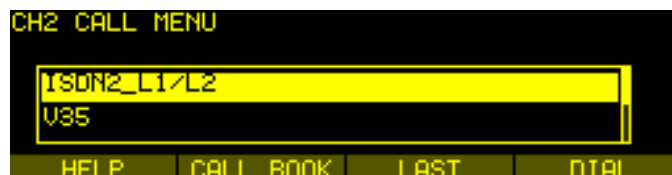
- <unit_unit>@<SIP_server_domain> in PROXY SIP mode (for instance., “phstr_xxx@sip.aeq.es”); when the unit you’re calling to is registered in the same SIP server as the one that’s calling, it’s enough just to use <name_unit>
- <unit_name>@<unit_IP_address> in DIRECT SIP mode (for instance, “phstr_xxx@192.168.1.84”).

IMPORTANT: the <unit_name> must be no longer than 19 characters.



CALL menu detail (IP-RTP mode)

- **INTERFACE:** allows you to define the communication interface.
 - For 64Kbps ISDN interface 1: ISDN1_L1, ISDN1_L2.
 - For 128Kbps ISDN interface 1: ISDN1_L1/L2.
 - For IP: Net1 (IP, associated to channel 1) or Net2 (IP, associated to channel 2).
 - For V35: V35.
 - When you want to configure an empty channel: NO INTERFACE.



INTERFACE menu detail

- **CODING / CODING (List):** lets you choose the encoding mode or profile.
 - **For IP** (IP interface in “PROXY SIP” or “DIRECT SIP” mode):
 - CODING (List):** you can select any of the lists created following the procedure described on section 3.1.10, for instance:
 - ONLY G722
 - LOW DELAY
 - HIGH QUALITY

- **For ISDN, V35 and IP** (IP interface in “RTP Point to Point” mode):

CODING: lets you choose the encoding algorithm among the available ones for the communication type. At the time this manual was written, the available algorithms are described below. We recommend consulting ANNEXE A to see a detailed description, as well as additional information on these encoding modes.

- OPUS (only for IP interface)
 - Mode: Mono / Stereo
 - Bit rate: 12Kbps / 20Kbps / 32Kbps / 64Kbps / 128Kbps / 192Kbps
 - Sampling rate: 48KHz
- G.711 A-Law mono
 - Mode: Mono
 - Bit rate: 64Kbps
- G.711 μ -Law mono
 - Mode: Mono
 - Bit rate: 64Kbps
- G.722
 - Mode: Mono
 - Bit rate: 64Kbps
- AEQ LD+
 - Mode: Mono / Stereo
 - Bit rate: 64Kbps / 128Kbps / 192Kbps / 256Kbps / 384Kbps
 - Sampling rate: 16KHz / 32KHz / 48KHz
- MPEG L2
 - Mode: Mono / Stereo / Dual / JStereo
 - Bit rate: 64Kbps / 128Kbps / 192Kbps / 256Kbps / 384Kbps
 - Sampling rate: 16KHz / 24KHz / 32KHz / 48KHz
- AAC-LC (AAC encoding modes are optional. Consult your authorized distributor)
 - Mode: Mono / Stereo / MS Stereo
 - Bit rate: 32Kbps / 64Kbps / 96Kbps / 128Kbps / 192Kbps / 256Kbps
 - Sampling rate: 24KHz / 32KHz / 48KHz
- AAC-LD (AAC encoding modes are optional. Consult your authorized distributor)
 - Mode: Mono / Stereo / MS Stereo
 - Bit rate: 32Kbps / 64Kbps / 96Kbps / 128Kbps / 192Kbps / 256Kbps
 - Sampling rate: 48KHz
- PCM
 - Mode: Mono / Stereo
 - Bit rate: between 384 and 2304Kbps
 - Sampling rate: 32KHz / 48KHz
 - Bits per sample: 12 (DAT) / 16 / 20 / 24
- Hybrid mode 1
 - TX: MPEGL2_64_48
 - RX: G722
- Hybrid mode 2
 - TX: G722
 - RX: MPEGL2_64_48
- AUTOSYNC: enables the auto-configuration of the encoding parameters according to the information in the received signal (only for ISDN and v35)

NOTE: For ISDN only the 64Kbps and 128Kbps modes are available (depending on the configuration of ISDN interface). For V35 all the 64, 128 and 256Kbps mode are available (depending on the frequency detected for V35 interface).

- **BACK UP:** enables you to configure the automatic backup option in case of connection failure. The menu and the options included are identical to the ones described previously in section 3.1.6.1 of this manual.

Fields that only appears for IP connections:

- **RX BUFFER MODE:** allows you to define the working mode of the reception buffer. This option will appear only when IP interface is selected (see section 3.1.6).
- **RX BUFFER (MAX):** indicates the maximum size of the reception buffer. This option will appear only when IP interface is selected.
- **RTP INTERFACE** (in “PROXY SIP” or “DIRECT SIP” mode): indicates the LAN port through which the audio streaming will be done if an IP type connection has been selected. This option will appear only when IP interface is selected and configured in “PROXY SIP” or “DIRECT SIP” mode.
- **LOCAL MEDIA PORT** (only in “RTP Point to Point” mode): value of the IP port selected for transmitting audio at origin over IP. Minimum value 1,024. Maximum value 65,534. This option will appear only when IP interface is selected and configured in “RTP Point to Point” mode.
- **DEST MEDIA PORT** (only in “RTP Point to Point” mode): value of the IP port selected for transmitting audio at destination over IP. Minimum value 1,024. Maximum value 65,534. This option will appear only when IP interface is selected and it's configured in “RTP Point to Point” mode.

The last line of the menu on the screen displays four options associated with the 4 contextual buttons:

- **HELP:** this feature is pending of development in future firmware versions.
- **CALL BOOK:** direct access to the internal phone book stored in the unit (see section 3.1.2 in this manual), where you can select a record and, using the DIAL option, make the call.
- **LAST:** accesses the record of the last ten calls made over the currently selected communication channel.
- **DIAL:** select this option to make a call to the terminal device whose data are visible in the CALL menu. You can get the same result by pressing the CALL button on the front panel a second time.

4. CONFIGURATION AND OPERATION OPTIONS DESCRIPTION. REMOTE CONTROL.

Configuration and operation of Phoenix STRATOS can be carried out remotely by means of the application “**AEQ ControlPHOENIX**” (AEQ Phoenix STUDIO, STRATOS, ALIO, MERCURY, VENUS and VENUS V2 Audiocoders Configuration and Control Software). **IMPORTANT NOTE:** For that purpose, the equipment must be previously configured to accept remote control connections (see **section 3.1.7** of this manual).

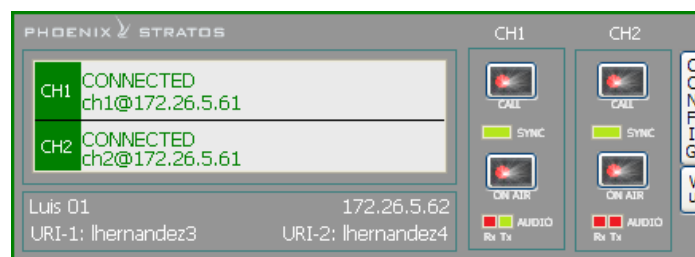
The version that is provided together with the equipment (**2.2.0.4** or higher) can control up to **2 units** per software instance. If you need to manage more than 2 Phoenix audiocoders at the same time, please contact AEQ sales department to purchase a multicodec-license for “AEQ ControlPHOENIX”.

Please have the user manual of this application at hand. Install and configure it and add the equipment to the controlled equipments list in order to follow the explanations in this and following chapters step by step. This manual will describe the technical characteristics of the most complex options only, while the detailed operation is detailed in “AEQ ControlPHOENIX” user’s manual.

4.1. Individual codec control window.

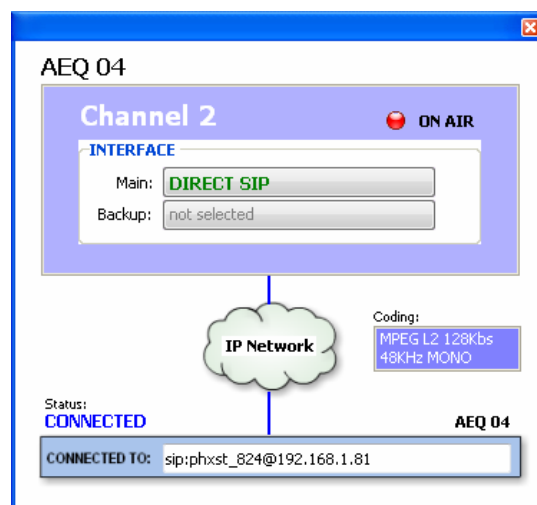
The individual codec control window is thoroughly described in chapter 6 of “AEQ ControlPHOENIX” user’s manual.

The screen corresponding to PHOENIX STRATOS is as follows:



The left zone shows the general status of both communications channels, CH1 and CH2 (CONNECTED, OK, REGISTERING, etc), as well as the remote equipment’s name in case it is connected. We can click in any of both areas in order to show a window that provides all the details of the channel we have clicked on.

- Channel status: **OFF AIR / ON AIR**.
- **INTERFACE:** indicates the interface configured for that channel (**IP, V35** or **ISDN**), its working mode (RTP Raw, DIRECT SIP or Proxy SIP for IP interface; L1 64k, L2 64k o L1+L2 128k for ISDN) and the backup interface (when defined).
- **Coding:** indicates the coding algorithm or profile: G711, G722, MPEG L2, AAC, <SIP CODEC PROFILE>... Check the complete coding algorithms list in section 1.3 of this manual and **APPENDIX A**. This section also indicates **binary rate** (128 Kbps for example), the **sampling frequency** (48KHz, for example) and the **mode** (Mono, Stereo, Dual, Jstereo or MS Stereo).
- **Status:** SIP registering status for IP connections using Proxy SIP mode (“REGISTERING”, “REGISTRATION_ERROR”, “REGISTERED”) or interface status (“OK”, “CONNECTED”, “CONNECTED_NO_DATA”).

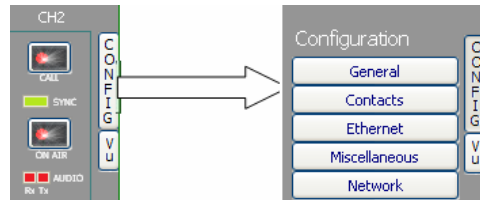


- **CONNECTED TO:** calling/called equipment's name or number (identifier, number or Unknown when ID is not provided) or "not connected" when there is no established communication.

The lower part of the window identifies each unit by its given name, IP address and each channel URIs.

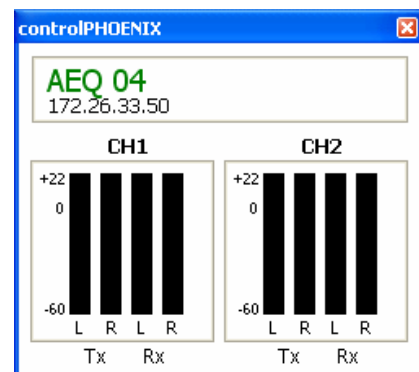
For both channels, CH1 and CH2, we can find "CALL" and "ON AIR" buttons/indicators at the right, as well as the "SYNC" indicator and audio presence indicators for both directions: transmission ("Tx") and reception ("Rx").

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



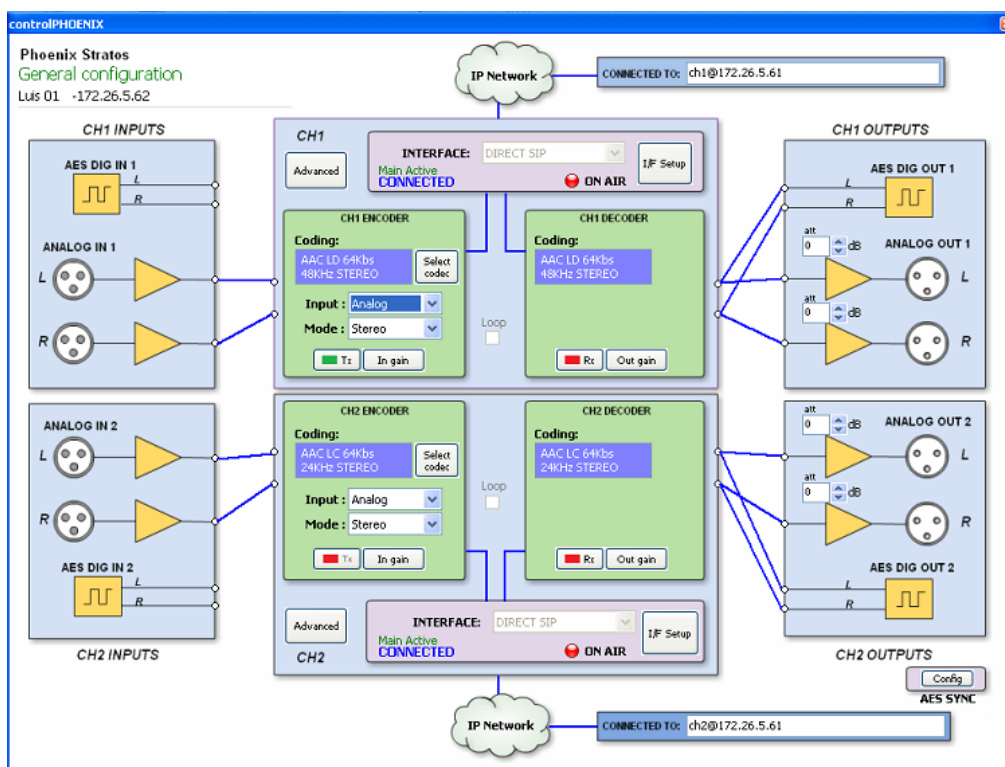
Just click on "CONFIG" button again in order to close this menu.

You can access to the equipment's remote vumeters by clicking on the "VU" button located below "CONFIG" one. They will appear into a floating window that you can move to the desired position. It is possible to open up to two vumeter windows (this is useful, for example, to check the correct audio transmission between two units that are connected and controlled by the application). If you try to open a third vumeters window, the first one will be closed. Click on the right top cross in order to close a vumeters window.



The represented vumeters correspond, for both channels, and from left to right, to audio transmitted to the channel (L and R) and received from it (also L and R).

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



4.2. IP interface connection modes.

In order to establish an IP communication using CH1 or CH2, first we need to choose one among three available connection modes within the “INTERFACE” drop-down menu of the desired channel: “PROXY SIP”, “DIRECT SIP” and “RTP Point to Point (RAW)”.

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

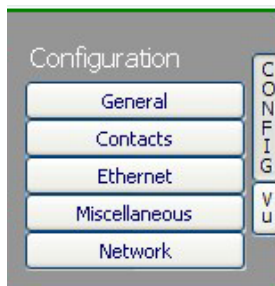
It is important to know the details of each type of connection, so they are explained below.

4.2.1. PROXY SIP.

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

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

- **<unit_name>**, for instance: “[phxst_231](#)” o “[phoenixMaster](#)”
- **<unit_name>@<realm_SIP_server>**, for instance: “[phxst_231@sip.aeq.es](#)” or “[phoenixMaster@sip.aeq.es](#)”
- **<unit_name>@< SIP_server_IP>**, for instance: “[phxst_231@232.168.1.2](#)” or “[phoenixMaster@232.168.1.2](#)” (where 232.168.1.2. is the AEQ’s SIP server “[sip.aeq.es](#)”)
- **<unit_name>@<SIP_server_IP>:<Port>**, when the SIP port of SIP server is not the 5060 (SIP Standard port) the identifier must include the used port, for instance: [phxst_231@sip.aeq.es:5061](#)



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

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

- In “**SIP Parameters**” submenu you can find:
 - **User Name:** enables you to edit the name of the unit and how it will be reflected in the diverse internal menus of the unit. For a start we recommend you not to change the configured default User Name, “[phxst_231](#)” for instance.
 - **Display Name:** editable name. This is the public name of the equipment that will be used in SIP server, so you can recognize the equipment with this identifier externally to the system.
 - **Proxy SIP Account:** enables you to select a Proxy SIP account from a previously created and stored list. In case an account is selected, the parameters described next would be automatically loaded (confirmation is requested).

- **Proxy Provider:** enables you to select the external SIP server with which the unit will work from a previously stored list. By default, AEQ server will be selected.
- **Authentication:** enables you to edit the password and security information for the user profile associated with the unit in the previously selected SIP server. By default, the data configured in this field in order to use AEQ server are the following:
 - o **User:** the “User Name” configured in Factory, “phstr_231” for instance.
 - o **Pwd:** the Password associated to that user.
 - o **Realm:** the domain where the SIP Server is located, by default: sip.aeq.es.

- You can find the NAT mode selection at “**NAT Traversal**” submenu.

NAT Traversal is a set of tools used by the equipment in order to surpass the NAT (Network Address Translation) performed by some routers. We can select among several modes depending on the kind of network the unit is connected to.

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

See more details in section 4.3 of this manual.

- The rest of options to be configured are:
 - o **FEC mode:** this option allows you to configure whether FEC (Forward Error Correction) is used or not (there is a trade-off for a bigger binary rate). See section 4.4.
 - o **Local media port:** this option allows you to configure the value of the IP port selected to transmit audio at origin over IP. Minimum value 1,024. Maximum value 65,534. Usually recommended value 5004.
 - o **Adaptive/Fixed** and **Adaptive buffer max/Fixed buffer length:** this option allows you to configure the type and maximum size of reception buffer. See section 4.4.
 - o **Symmetric RTP:** this option allows you to force the local unit to send audio to the same IP and port from which it is receiving audio. The destination port specified when making the call will be ignored when we receive packets from the remote equipment. This option will allow you to connect to an equipment with unknown IP and/or port (because it's behind a router with NAT, for instance).

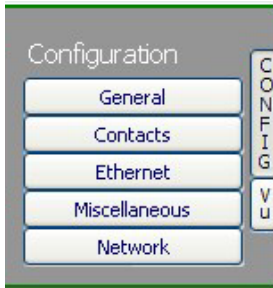
Each unit will send audio to the “Local media port” of the remote equipment automatically, thanks to the SIP signaling protocol. That signaling also accomplishes coding profile negotiation and call establishment / release from any of both sides of the communication once the remote equipment has been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

4.2.2. DIRECT SIP.

This type of connection is selected when you have a connection with SIP protocol in the signaling phase prior to connection but without the presence of an external SIP server. It is necessary to know the IP address of the equipment you want to call in advance.

In order to call in Direct SIP mode, you must take into account that for the **URI** or SIP identifier of the equipment the right syntax is “<unit_name>@<unit_IP_address>” type (for instance, “[phstr_231@172.26.5.57](#)”).

When the correspondent **SIP port** is not the 5060 (SIP Standard port) the identifier must include the used port. For instance: “[phxstr_231@172.25.32.11:5061](#)”.



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

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

- In “**SIP Parameters**“ submenu you can find:
 - **User Name:** enables you to edit the name of the unit and how it will be reflected in the diverse internal menus of the unit. For a start we recommend you not to change the configured default User Name, “phxst_231” for instance.
 - **Display Name:** editable name. This is the public name of the equipment, so you can recognize the equipment with this identifier externally to the system.

- You can find the NAT mode selection at “**NAT Traversal**“ submenu.
 NAT Traversal is a set of tools used by the equipment in order to surpass the NAT (Network Address Translation) performed by some routers. We can select among several modes depending on the kind of network the unit is connected to.

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

See more details in section 4.3 of this manual.

- The rest of options to be configured are:
 - **FEC mode:** this option allows you to configure whether FEC (Forward Error Correction) is used or not (there is a trade-off for a bigger binary rate). See section 4.4.
 - **Local media port:** this option allows you to configure the value of the IP port selected to transmit audio at origin over IP. Minimum value 1,024. Maximum value 65,534. Usually recommended value 5004.
 - **Adaptive/Fixed** and **Adaptive buffer max/Fixed buffer length:** this option allows you to configure the type and maximum size of reception buffer. See section 4.4.
 - **Symmetric RTP:** this option allows you to force the local unit to send audio to the same IP and port from which it is receiving audio. The destination port specified when making the call will be ignored when we receive packets from the remote equipment. This option will allow you to connect to an equipment with unknown IP and/or port (because it’s behind a router with NAT, for instance).

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

4.2.3. RTP Point to Point (RAW).

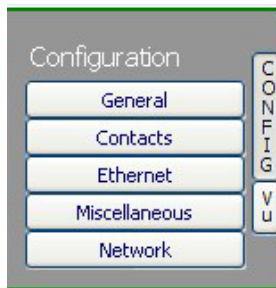
This type of connection is selected when the connection over IP will be an RTP type link with calling of the IP address to IP address type. Obviously there is no advanced signaling protocol in this scenario and you will need to established, set parameters and disconnect communication from both ends. Audio encoding must be the same (and explicitly specified) in both ends of the communication.

In order to avoid that hassle, making the calling, hanging up and coding selection tasks easier, as it will be necessary to do it from one end only, “**SmartRTP**” can be activated in both involved audiocoders (provided that they are AEQ Phoenix compatible with this mode).

If the operating mode required for a contact is “**RTP raw**”, the only valid equipment identifier is:

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

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



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

You can access the IP configuration submenu for RTP Raw mode by clicking on “**IF Setup**” button, and that it is explained in section 6.1.4.3 of “AEQ ControlPHOENIX” user’s manual.

The parameters to be configured are:

- **FEC mode:** this option allows you to configure whether FEC (Forward Error Correction) is used or not (there is a trade-off for a bigger binary rate). See section 3.4.
- **Local media port:** this option allows you to configure the value of the IP port selected to transmit audio at origin over IP. Minimum value 1,024. Maximum value 65,534. Usually recommended value 5004.
- **Adaptive/Fixed and Adaptive buffer max/Fixed buffer length:** this option allows you to configure the type and maximum size of reception buffer. See section 3.4.
- **Symmetric RTP:** when “**SmartRTP**” mode is not activated, this advanced option at least allows the user to force the local unit to send audio to the same IP and port from which it is receiving audio. The destination port specified when making the call will be ignored when we receive packets from the remote equipment. This option will allow you to connect to an audiocoder with unknown IP and/or port (because it’s behind a router with NAT, for instance).

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

4.2.3.1. Sending audio to multiple destinations: Broadcast, Multicast and Multiple Unicast.

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

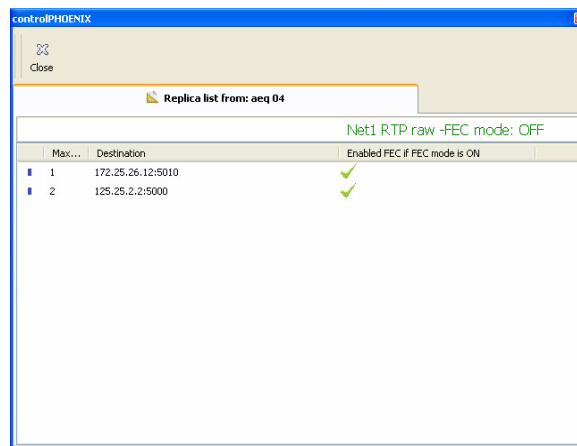
a) Broadcast: the audio stream can be sent to all the equipments within a local network, only by specifying a special address in the destination address field. This address is calculated as the network address with the equipment part filled with 1's. For instance: if the IP address of our codec is 192.168.20.3 and network mask is 255.255.255.0, the corresponding broadcast address is 192.168.20.255. However, if the network mask was, for example, 255.255.0.0, then the broadcast address would be 192.168.255.255. The audio will be sent to a given port, so the receiving equipments should have "local media port" set up to this same port so they are able to receive the RTP stream.

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

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

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

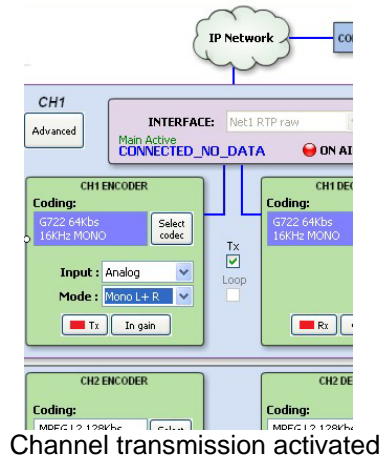
The "**Multiple unicast**" button (within the IP interface configuration screen Net1 or Net2) gives access to an information window where the replicas that are active at a certain moment are shown.



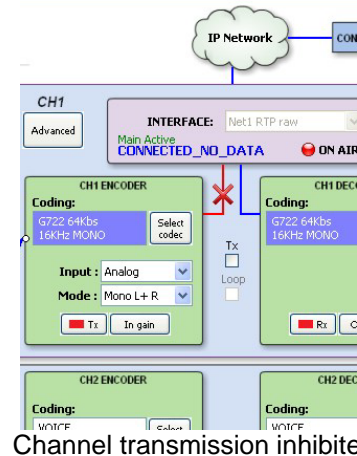
You can add replicas to a RTP call (or edit the existing ones) by previously creating a contact that includes them (or by editing a existing contact). See section 5.1.7.1 of of "AEQ ControlPHOENIX" user's manual.

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

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



Channel transmission activated



Channel transmission inhibited

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

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

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

4.3. NAT TRAVERSAL.

NAT Traversal is a set of tools used by the equipment in order to surpass the NAT (Network Address Translation) performed by some routers. We can select among several modes depending on the kind of network the unit is connected to.

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

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

Due to the technical complexity inherent in most of the parameters involved in this NAT TRAVERSAL menu and the importance that any modification has in the final operation of the unit, we recommend that only highly qualified personnel in possession of all the technical documentation and manuals work on this NAT configuration menu. For additional information, see **APPENDIX C5**.

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

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

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

The unit uses no mechanism to traverse devices with NAT. This mode will be used only to operate in the local network (all of the SIP participants are in the same local network, including the Proxy SIP, if we use it).

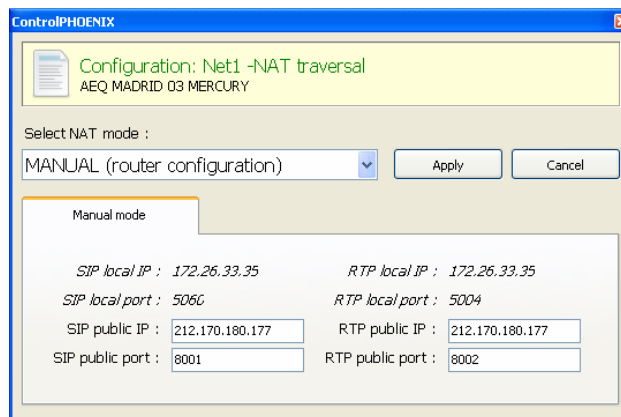
4.3.2. Manual NAT: "MANUAL (router configuration)".

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

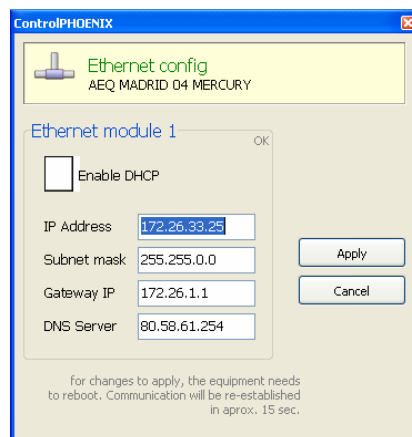
The procedure is described in detail in the following Application Note (we recommend you to read it when you decide to use this working mode):

AEQ PHOENIX AUDIOCODECS. APPLICATION NOTE 0 - C (Adapted to STRATOS)
 (Connecting AEQ Phoenix units through Internet. Complex scenario configuration. Through LOCAL network(s), DHCP not used, manual NAT. Making use of AEQ Proxy SIP).

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



1. **SIP LOCAL IP:** read-only parameter that tells you the IP of the IP interface of the unit as regards SIP, so that the latter can, in turn, convey this to the router or firewall administrator when it is configured. For instance [172.26.33.35](#). It can be set in order to adapt it to network necessities in menu: "Configuration" → "Ethernet".



2. **SIP LOCAL PORT:** read-only parameter that tells you the port of the IP interface of the unit used for SIP signaling, so that the latter can, in turn, convey this to the router or firewall administrator when it is configured. Before checking the value of this parameter you should have configured previously whether you want to work with Proxy or not and restart the unit.
3. **SIP PUBLIC IP:** parameter that will tell the unit which public IP will correspond to it, so that it can include the said IP in its SIP messages. The router or firewall administrator must tell you the value of this parameter. For instance [212.170.180.177](#)
4. **SIP PUBLIC PORT:** parameter that will tell Phoenix which public port it will have corresponding to its local SIP port. The router or firewall administrator must tell you the value of this parameter in order to make the required port-forwarding. For instance: 8001.
5. **RTP LOCAL IP:** read-only parameter that tells you the IP of the IP interface of the unit as regards RTP, so that it can, in turn, convey this to the router or firewall administrator when it is configured. You will usually configure the same network interface as for SIP, so it will be the one configured in point number 1: for instance [172.26.33.50](#)
6. **RTP LOCAL PORT:** read-only parameter that tells you the port of the IP interface of the unit as regards RTP, so that the latter can, in turn, convey this to the router or firewall administrator when it is configured. Usually the shown port is 5004.
7. **RTP PUBLIC IP:** parameter that will tell the unit which public IP will correspond to the RTP of its IP interface, so that it can send the said IP in its SIP messages. The router or firewall administrator must tell you the value of this parameter. Usually the administrator will take out SIP traffic and RTP using the same public IP configure in point number 3. For instance [212.170.180.177](#)
8. **RTP PUBLIC PORT:** parameter that will tell the Phoenix STRATOS which public port will correspond to the RTP of its IP interface, so that it can send the said port in its SIP messages. The router or firewall administrator must tell you the value of this parameter in order to make the required port-forwarding. For instance: 8002

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

4.3.3. AUTO 1 (local network audio).

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

4.3.4. AUTO 2 (local network audio).

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

4.3.5. AUTO 3 (audio over internet).

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

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

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

4.3.6. AUTO 4 (audio over internet).

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

4.4. FEC modes and reception buffer configuration.

- **FEC error correction mode.** Error correction is performed by sending redundant information that allows the receiver to recompose the lost data in case of not-perfect transmissions.

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

- **LOWEST:** generates a 40% higher binary rate and produces a 575ms additional delay.
 - **LOW:** generates a 50% higher binary rate and produces a 375ms additional delay.
 - **MIDDLE:** generates a 66% higher binary rate and produces 225ms additional delay.
 - **HIGH:** duals the binary rate producing 125 ms additional delay.
- **Adaptive/Fixed:** you can set up the **reception buffer** as adaptive or fixed. In the first case, its size will vary according to the network transmission quality. In fixed mode, its size will be steady according to manual configuration.
 - **Adaptive Buffer Max/Fixed buffer length:** this is the maximum size of the reception buffer. When it is defined as adaptive, Phoenix STRATOS will start to shorten it from this value as the network's transmission quality allows. If it is defined as FIXED, this max value will remain, as the buffer's size won't be varied during the connection. This value must be set in milliseconds. The longer the buffer is, packet misses will be less likely, but base delay will also be longer, especially if the buffer is set to FIXED mode.

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

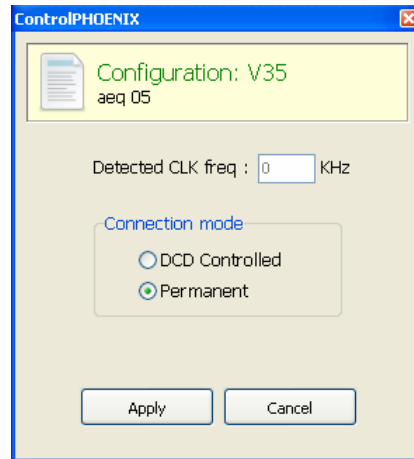
In high-quality PCM connections, you can start using highest quality modes (48KHz, 24 bits, mono or stereo only if required), and if you can't obtain the desired quality and/or stability (no noises present) and good delay, you can lower quality progressively until, for example, 16 bit (CD quality audio).

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

4.5. V35 interface configuration.

In order to establish a V35 communication using CH1 or CH2, first we need to choose V35 option within the “INTERFACE” drop-down menu of the desired channel.

We can access the V35 configuration submenu by clicking on “I/F Setup”. This menu is described in sections 6.1.4.1 of “AEQ ControlPHOENIX” user’s manual. V35 interface configuration is limited to selection of the operating mode (permanent or DCD signal controlled) and an informative window that displays the frequency detected for the incoming clock.



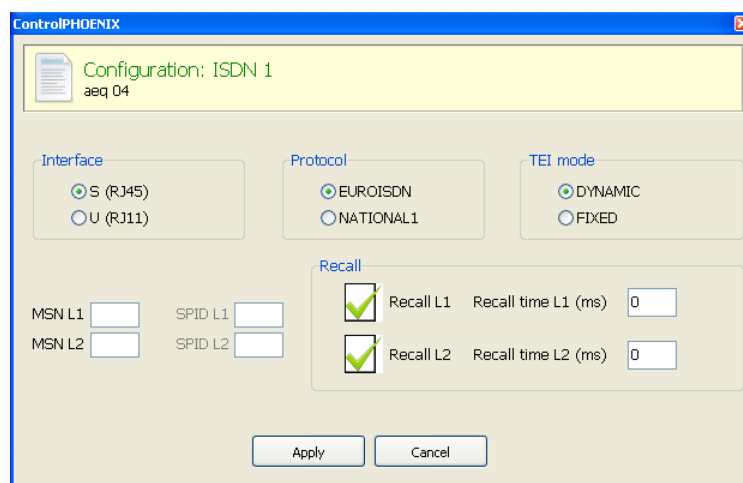
4.6. ISDN (ISDN1) communications interface configuration.

There is an ISDN interface installed into **Phoenix STRATOS** (ISDN1).

In order to establish an ISDN communication using CH1 or CH2, first we need to choose within the “INTERFACE” drop-down menu of the desired channel one of the following options: “ISDN1 L1+L2 128k”, “ISDN1 L1 64k” or “ISDN1 L2 64k”.

We can access the ISDN configuration submenu by clicking on “I/F Setup”. This menu is described in section 6.1.4.4 of “AEQ ControlPHOENIX” user’s manual.

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

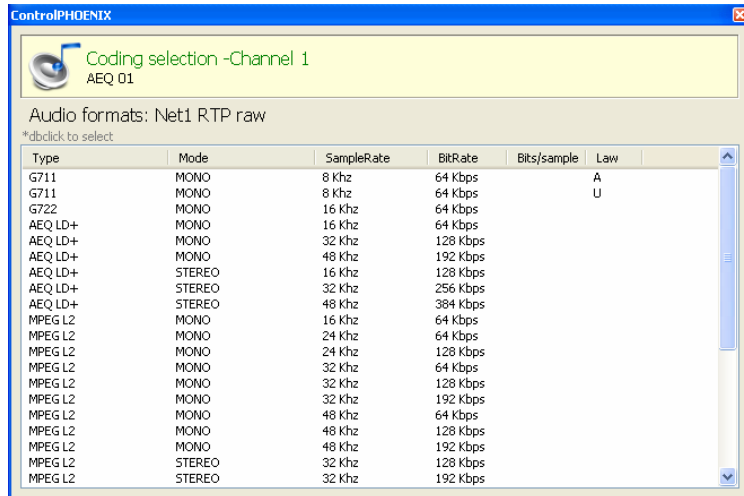


4.7. Coding selection.

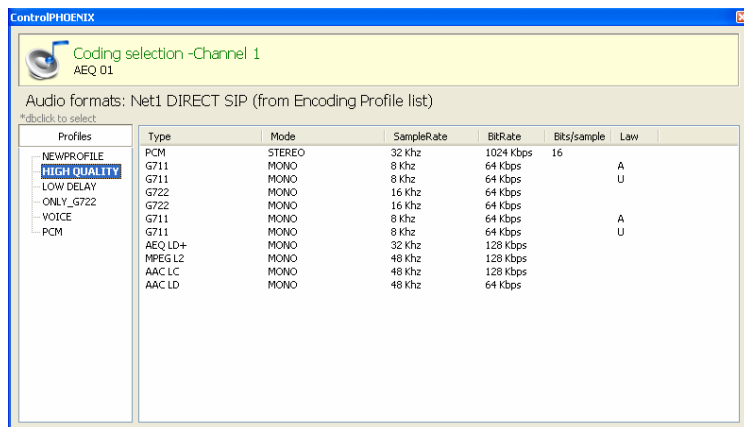
See section 6.1.3.1 ("Coding selection") at "AEQ ControlPHOENIX" user's manual.

By clicking on the "Select codec" button located inside the "ENCODER" area of the general configuration window, you can access the coding selection screen. In this window, with "INTERFACE" is configured as **V35, ISDN or IP** (set as "RTP raw"), only the modes that are compatible with the communications interface selected appear.

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



However, when the interface is **IP** and configured in any of the **SIP** modes ("DIRECT SIP" or "PROXY SIP"), particular coding algorithms won't be selected and connection "profiles" will be defined instead, containing one or more modes. This is like that because SIP allows the participants in a communication to negotiate the coding algorithm, so the one to be finally used will be limited to the list the caller presents.



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

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

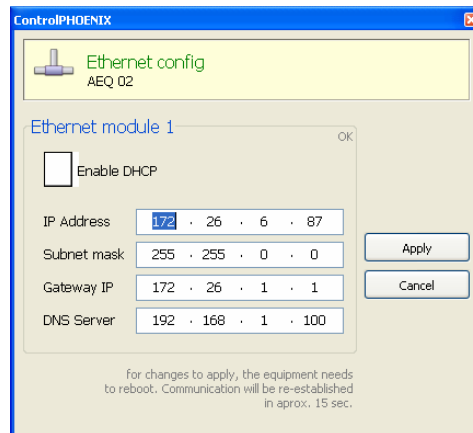
Each one of the stored entries includes an alphanumeric identifier and a list of algorithms to use, organized in order of preference. There several preset profiles in the unit, grouped by delay, quality... etc.

Profiles can be added, modified or deleted in the "Encoding Profile Management (SIP)" screen, accessible from the "Tools" menu in the upper Menu bar, described in paragraph 5.1.8 of "AEQ ControlPHOENIX" user's manual.



4.8. Ethernet Port configuration.

The “**Ethernet config**” menu allows you to configure the IP parameters of the Ethernet interface in the unit.



The parameters to be configured are:

- **Enable DHCP:** enables the activation or deactivation of the automatic configuration of IP addresses, masks and gateways. There must be a DHCP server in the network the unit is connected to in order to make this option work. When “Enable DHCP” is validated, the following parameters will be filled automatically; when “Enable DHCP” is not validated you will be able to change them manually.
- **IP Address:** valid IP address associated with that interface.
- **Subnet mask:** valid subnet mask associated with that interface.
- **Gateway IP:** valid gateway or network gateway address associated with that interface.
- **DNS Server:** IP address of the external addresses resolution server, valid in the geographic zone where codec is placed, or internal server (inside the local network) authorized to translate alphanumeric URL identifiers into IP addresses.

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

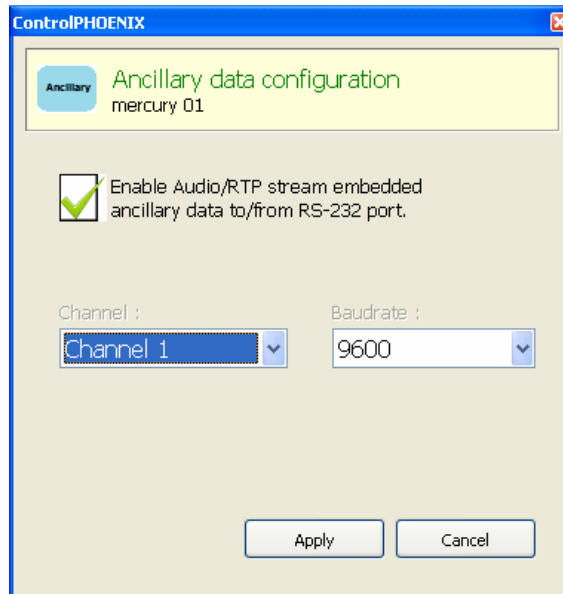
If you have any doubts, please consult your IT network technician or directly contact the AEQ or authorized distributors’ technical support department.

4.9. RS-232 Ancillary (End-to-end data channel).

This option is available in “Configuration” → “Miscellaneous”.

PHOENIX STRATOS allows you to perform a bidirectional data transmission between units by means of RS232 rear connector. This option may be useful in order to send files from one location to another, or also to control remotely other units with RS232 serial interface. Ancillary data travels through IP in a separate stream from the audio.

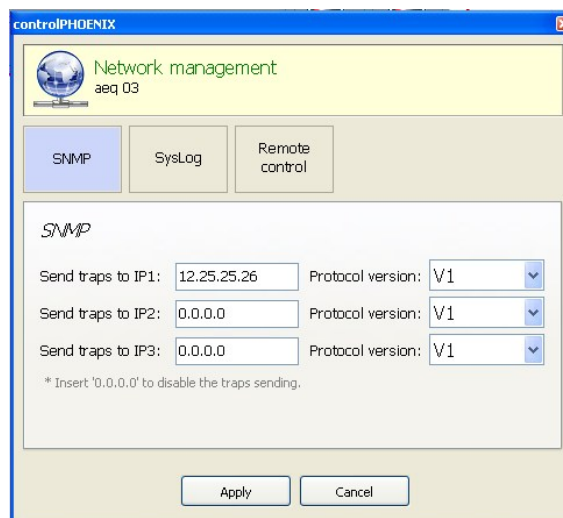
This option allows you to activate/deactivate that data transmission, associate it to channel 1 or 2 and select the binary rate. Binary rates of 9600, 19200 or 38400 bauds can be selected (only 9600 bauds in case the selected channel is transmitted through an ISDN connection).



4.10. SNMP configuration.

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

You can access the configuration menu in “Configuration” → “Network”.



AEQ PHOENIX Audiocodex (Mercury, Venus, Studio and STRATOS) can connect to **up to 3 SNMP clients**, installed in remote PCs, by simply configuring their IP addresses in the “SNMP” tab of previous menu. Once one or more SNMP clients are connected and the corresponding “MIB” descriptive file has been loaded (it can be downloaded from the equipment's Web Interface: see section 7.8 of this manual), the audiocodex will send/accept different types of informations to/from each client:

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

- phxCh1NoTxAudioAlarm (Audio detection event at Channel 1's input)
- phxCh1NoRxAudioAlarm (Audio detection event at Channel 1's output)
- phxCh2NoTxAudioAlarm (Audio detection event at Channel 2's input)
- phxCh2NoRxAudioAlarm (Audio detection event at Channel 2's output)
- phxCh1NoAudioSyncAlarm (Sync event at Channel 1 while connected)
- phxCh2NoAudioSyncAlarm (Sync event at Channel 2 while connected)
- phxCh1CallEndAlarm (Call ended on Channel 1 due to incoming RTP traffic loss)
- phxCh2CallEndAlarm (Call ended on Channel 2 due to incoming RTP traffic loss)
- phxOtherAlarm (Other alarms, see name) -> Fail to register in SIP PROXY server, activation / deactivation of BACKUP interface for Ch1 and/or Ch2.
- coldStart alarm (Starting from unit off. This is an standard SNMP alarm, it appears only one time and has no activation or deactivation).

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

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

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

- phxCh1TxAudioThreshold (Audio threshold for channel 1's input)
- phxCh1TxAudioInterval (Audio interval for channel 1's input)
- phxCh1RxAudioThreshold (Audio threshold for channel 1's output)
- phxCh1RxAudioInterval (Audio interval for channel 1's output)
- phxCh2TxAudioThreshold (Audio threshold for channel 2's input)
- phxCh2TxAudioInterval (Audio interval for channel 2's input)
- phxCh2RxAudioThreshold (Audio threshold for channel 2's output)
- phxCh2RxAudioInterval (Audio interval for channel 2's output)

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

- phxCh1Status (Channel 1's status) --> (OK, CONNECTED, CONNECTING, DISCONNECTING, CALLING, NO LINK, REGISTER ERROR, REGISTERING, CALL ERROR)
- phxCh2Status (Channel 2's status) --> (OK, CONNECTED, CONNECTING, DISCONNECTING, CALLING, NO LINK, REGISTER ERROR, REGISTERING, CALL ERROR)
- phxCh1TxAudio (Ch1's audio input indicator status). Indicates the status of the virtual LED indicating audio presence, according to the corresponding configured parameters.
- phxCh1RxAudio (Ch1's audio output indicator status). Indicates the status of the virtual LED indicating audio presence, according to the corresponding configured parameters.
- phxCh2TxAudio (Ch2's audio input indicator status). Indicates the status of the virtual LED indicating audio presence, according to the corresponding configured parameters.
- phxCh2RxAudio (Ch2's audio output indicator status). Indicates the status of the virtual LED indicating audio presence, according to the corresponding configured parameters.
- phxCh1OnAir (Ch1's ON AIR option activated or not)

- phxCh2OnAir (Ch2's ON AIR option activated or not)
- phxCh1Synced (Ch1's audio synchronized or not)
- phxCh2Synced (Ch2's audio synchronized or not)
- phxCh1BackupInterfaceActive (Whether or not Ch1's backup interface is being used on a call)
- phxCh2BackupInterfaceActive (Whether or not Ch2's backup interface is being used on a call)
- mib2.system (sysUpTime, sysContact, sysDescr, sysServices...): Standard SNMP commands indicating things such as equipment's turn on time, etc. For more information, please check MIB-II specification in RFC1213 (<http://tools.ietf.org/html/rfc1213>)

For more information, please consult section 7.8 of this manual and section 6.5.1 of "AEQ ControlPHOENIX" application manual.

5. QUICKSTART USER'S GUIDE. MANUAL CONTROL.

To gain a complete grasp of the PHOENIX STRATOS, we **recommend** reading the previous chapters carefully. The paragraphs below describe the basic actions you will need to take to **manually** operate the equipment. If you need more detail, review the information given in the previous sections of this manual.

5.1. PHOENIX STRATOS connections.

5.1.1. Power supply.

The 3-pin IEC power feed connectors are located on the back panel of the codec. Connecting one is enough. You can use both for redundancy if you have two different power supplies.

5.1.2. Audio.

Section 2.2 of this manual gives a detailed description of the physical connection components of the equipment. A simple approach to the connection procedure follows:

- Connect the line inputs in analog or digital format that you need, according to the description of the connectors installed in the PHOENIX STRATOS rear panel. XLR connectors for the analog inputs and DB15 connector for the digital inputs.
- From **MAIN STATUS→CH1 STATUS→AUDIO IN SOURCE**, select the type of signal to be used for channel 1: ANALOG or DIGITAL. Repeat this step for channel 2.
- Connect the line outputs in analog or digital format that you need, according to the description of the connectors installed in the PHOENIX STRATOS rear panel. XLR connectors for the analog outputs and DB15 connector for the digital outputs.
- There is no need to define the output type because the PHOENIX STRATOS offers, by default, all the outputs duplicated in analog and digital format.
- If you need to use headphones for local monitoring, the ¼" Jack with its volume control is available in the right-hand area of the front panel.
- The configuration of the signal to monitor in this headphones output is performed in the **MAIN STATUS→HPHONES** menu.

5.1.3. Communication interfaces.

Depending on the type of communication, a certain type of connection will be needed:

- **IP connection:** Connect Ethernet cable (CAT5 or higher) terminated in an RJ45 10/100 BT connector to the connector installed for this purpose in the rear panel of the unit. The selected cable will be the direct 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 2.2.7 of this manual.
- **ISDN connection:** Connect standard cable terminated in an RJ11 or RJ45 connector to one of the two connectors made available for this purpose in the rear section of the ISDN module, according to the protocol and standard to be used (see section 2.2.8).
- **V.35 connection:** Connect standard cable terminated in a DB25 connector to the connector installed for this purpose in the rear section of the unit (see section 2.2.5).

5.2. Powering up the codec.

- Once the codec has been connected to the mains, after about 15 seconds, the OLED screen in the display will turn on, showing the AEQ logo and the name of the unit.



- After a few seconds, the MAIN STATUS menu screen will be displayed. Check to see that there is no ERROR message (the screen has dimmed lighting, press any key to see it properly).



- The unit is ready to be used.

5.3. Establishing an ISDN communication.

- Ensure that the equipment is powered up.
- Check that the RJ45 or RJ11 cable is correctly connected and secured.
- Verify the status of the ISDN communications module. Ensure that no error message appears in the MAIN STATUS menu screen for this communications module.
- Establish the appropriate audio configuration (analog / digital input mode, gains, routing of inputs to the codec, suitable encoding, analog output modes).
- Set the VU meters at the most suitable adjustment point and check the input audio.
- Assign the ISDN interface to be used for the connection. For 64Kbps calls, all the B channels available will appear independently.
- Press the **CALL** button associated with the communications channel that will use the ISDN interface as the connection base; this action will open the connection configuration menu:
 - Check that the assigned ISDN interface is the desired one.
 - Select the encoding mode. Make sure that the mode you select is compatible with the communications interface selected, especially as regards the associated bit rates.
 - You have the possibility of using the backup option, with its corresponding configuration in **BACK UP**:
 - Activate or not the backup option.
 - Select the interface to use as a backup.
 - Select the encoding mode. Make sure that the mode you select is compatible with the communications interface selected, especially as regards the associated bit rates.
 - Define the minimum time interval for the connection to be considered down and the communication interface switching to be performed.
 - Select the number of retries to be made before considering the communication down.
 - Define the minimum time interval for the connection to be considered down if synchronization is not performed, and for the communication interface switching to be performed.
 - Define the destination toward which the backup connection is to be launched. Depending on the communication interface you have

- selected as a backup, this destination will be defined by an ISDN subscriber number, a SIP identifier (URI) or an IP address.
 - Enter the number of the contact to be called, either manually or from the internal phone book in **CALL BOOK**.
- Press the **CALL** button on the front panel again, or select the **DIAL** option on screen, to make the call.

If you select the number to call from the phone book, you can perform the same action by using the contextual button associated with the **DIAL** option.
- You can monitor the status of the call in the OLED screen:
 - **CALLING**.
 - **CONNECTING** (depending on the communication interface, this may be an extremely brief status).
 - **SYNCHRONIZING** (depending on the communication interface, this may be an extremely brief status).
 - **CONNECTED**.



Detail of Phoenix STRATOS calling

- Verify that the **SYNC** LED beneath the **CALL** button is lighted in green to indicate that the communication has been successfully established.
- Press the **ON AIR** button associated with that communications channel to enable audio to be transmitted over that interface in full duplex mode.
- Once the connection has been established with the remote end, confirm the presence of transmitted and received audio by checking the VU meters on the Phoenix STRATOS front panel.
 - If necessary, adjust the levels in **AUDIO SETTINGS** → **CODEC CHANNEL1 SETTINGS** (or **CODEC CHANNEL2 SETTINGS**) menu or in **ANALOG AUDIO SETTINGS** menu, according to the desired signal type.

5.3.1. Finalizing an ISDN communication.

- To finalize the communication, simply press the **CALL** button associated with the communications currently being used on the unit's front panel.

5.3.2. Reception and acceptance of an ISDN communication.

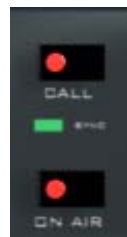
If the ISDN interface of the codec is suitably configured and the automatic answering mode is OFF, when a call is received:

- The unit will emit an audible notification signal. This signal can be disabled in **MENU SYSTEM** → **8-SETTINGS** → **BUZZER**.
- The red **CALL** button light corresponding to the communications channel that receives the call will blink simultaneously to notify you.
- The caller's information will appear on the OLED screen:
 - The communications interface through which the call is being received.
 - The communications channel through which the call is being received. The channels and interfaces are associated in **MENU SYSTEM** → **6-CHANNELS** → **Chn1** or **Chn2** → **INTERFACE**.
 - The calling number identifier.



Detail of the Phoenix STRATOS receiving a call

- To accept the call, press the **contextual button ACCEPT_C1** (when the call is received through communications channel 1) or **ACCEPT_C2** (when the call is received through communications channel 2).
You can also accept the call by pressing the **CALL** button associated with the communications channel that is receiving the incoming call.
- You can monitor the status of the call in the OLED screen:
 - **CONNECTING** (depending on the communication interface, this may be an extremely brief status).
 - **SYNCHRONIZING** (depending on the communication interface, this may be an extremely brief status).
 - **CONNECTED.**
- Verify that the **SYNC** LED beneath the **CALL** button is lighted in green to indicate that the communication has been successfully established.
- Press the **ON AIR** button associated with that communications channel to enable audio to be transmitted over that interface in full duplex mode.



- Once the connection has been established with the remote end, confirm the presence of transmitted and received audio by checking the VU meters on the Phoenix STRATOS front panel.
 - If necessary, adjust the levels in **AUDIO SETTINGS** → **CODEC CHANNEL1 SETTINGS** (or **CODEC CHANNEL2 SETTINGS**) menu or in **ANALOG AUDIO SETTINGS** menu, according to the desired signal type.

The automatic answering mode can be activated or deactivated in **MENU SYSTEM** → **6-CHANNELS** → **Chn1** or **Chn2** → **OFF HOOK**.

5.4. Establishing a V.35 communication.

- Ensure that the equipment is powered up.
- Check that the cable is properly connected and secured to the DB25 connector installed in the back panel of the codec.
- Verify the status of the V.35 communications module. Ensure that no error message appears in the MAIN STATUS menu screen for this communications interface. The unit must detect the frequency of the connected master clock, showing it in the previously mentioned screen.
- Establish the appropriate audio configuration (analog / digital input mode, gains, routing of inputs to the codec, suitable encoding, analog output modes).
- Set the VU meters at the most suitable adjustment point and check the input audio.
- Assign the V.35 interface to the channel.
- Press the **CALL** button associated with the communications channel that will use the V.35 interface as the connection base; this action will open the connection configuration menu.

- Check that the V.35 interface is assigned to the channel.
- Select the encoding mode.
- You have the possibility of using the backup option, with its corresponding configuration in **BACK UP**:
 - Activate or not the backup option.
 - Select the interface to use as a backup.
 - Select the encoding mode. Make sure that the mode you select is compatible with the communications interface selected, especially as regards the associated bit rates.
 - Define the minimum time interval for the connection to be considered down and the communication interface switching to be performed.
 - Select the number of retries to be made before considering the communication down.
 - Define the minimum time interval for the connection to be considered down if synchronization is not performed, and for the communication interface switching to be performed.
 - Define the destination toward which the backup connection is to be launched. Depending on the communication interface you have selected as a backup, this destination will be defined by an ISDN subscriber number, a SIP identifier (URI) or an IP address.
- Press the **CALL** button on the front panel again, or select the **DIAL** option on screen, to make the call.
- You can monitor the status of the call in the OLED screen:
 - **CALLING.**
 - **SYNCHRONIZING** (depending on the communication interface, this may be an extremely brief status).
 - **CONNECTED.**
- Repeat the process above in the unit at the other end.
- Verify that the **SYNC** LED beneath the CALL button is lighted in green to indicate that the communication has been successfully established.
- Press the **ON AIR** button associated with that communications channel to enable audio to be transmitted over that interface in full duplex mode.
- Once the connection has been established with the remote end, confirm the presence of transmitted and received audio by checking the VUmeters on the front panel.
 - If necessary, adjust the levels in **AUDIO SETTINGS→CODEC CHANNEL1 SETTINGS** (or **CODEC CHANNEL2 SETTINGS**) menu or in **ANALOG AUDIO SETTINGS** menu, according to the desired signal type.

5.4.1. Finalizing a V.35 communication.

- To finalize the communication, simply press the **CALL** button associated with the communications currently being used on the unit's front panel. Repeat the process in the unit at the other end.

5.5. Establishing an IP communication.

Several operating modes are available, depending on the protocol used for the communication initiation.

The simplest –yet still very effective- mode, that we recommend for a start, is IP mode (RTP Raw) with the “**SmartRTP**” option activated (this option can be found under the AUTO menu), as it will be adequate for most cases, together with the OPUS family of encoding algorithms.

When operating with audiocoders from other manufacturers, you can use either IP mode without “**SmartRTP**” (almost every codec out there can operate in RTP mode) or, alternatively, a SIP-based mode. When the IP address of the counterpart unit is not known (mobile connections, dynamic IP, etc) the PROXY SIP mode is recommended, as it relies on an external server to resolve IP addresses. On the other hand, if IP and ports are well-defined, it is better to use DIRECT SIP. Select any of the N/ACIP recommended encoding algorithms, such as G722 or MPEG-1 /2, depending on the quality required, acceptable delay and available bandwidth.

5.5.1. Establishing an IP communication in RTP mode using *SmartRTP*.

- Ensure that the equipment is powered up.
- Check that the RJ45 cable is correctly connected and secured.
- Check that the amber indicator lamp integrated into the communications module is active.
- Verify the status of the communications interface in the internal menu. Ensure that no error message appears in the MAIN STATUS menu screen for this communications module.
- Establish the appropriate audio configuration (analog / digital input mode, gains, routing of inputs to the codec, suitable encoding, analog output modes).
- Set the VU meters at the most suitable adjustment point and check the input audio.
- Enter the interface configuration and select the “RTP Point to Point” mode.
- Select the “LOCAL MEDIA PORT” and “DEST MEDIA PORT”: local IP port (through which the RTP audio is received) and destination IP port (to which the audio will be sent). Both ports may be the same.
- Configure the “**SmartRTP MODE**” option as “**ON**”.
- Assign the IP interface to the channel.
- Press the **CALL** button associated with the communications channel that is going to use the IP interface as the connection base; this action will open the connection configuration menu. Basically, ensure that all the fields are correct, and that they correspond to what you have configured previously:
 - Check that the assigned IP interface is the desired one.
 - Select the encoding profile to be used in the communication.
 - Select the working mode and the reception buffer size (see paragraph 4.4 of this manual in order to select the optimal buffer configuration depending on your application).
 - You have the possibility of using the backup option, with its corresponding configuration submenu in **BACK UP**:
 - Activate or not the backup option.
 - Select the interface to use as a backup.
 - Select the encoding mode. Make sure that the mode you select is compatible with the communications interface selected, especially as regards the associated bit rates.
 - Define the minimum time interval for the connection to be considered down and the communication interface switching to be performed.
 - Select the number of retries to be made before considering the communication down.
 - Define the minimum time interval for the connection to be considered down if synchronization is not performed, and for the communication interface switching to be performed.
 - Define the destination toward which the backup connection is to be launched. Depending on the communication interface you have selected as a backup, this destination will be defined by an ISDN subscriber number, a SIP identifier (URI) or an IP address.
 - Enter manually the IP address to be called.
- Press the **CALL** button on the front panel again, or select the **DIAL** option on screen, to make the call.
- Repeat the audio and mode configuration process in the other end. As “*SmartRTP*” option is active, the other codec will automatically connect as soon as it starts receiving RTP traffic, and so there will be no need to dial IP addresses or ports, accept the incoming call or even configuring the encoding algorithm. Only when “*SmartRTP*” is not active, you’ll have to repeat the calling process at the other end (because RTP is connection-oriented, you also need to connect to the IP of the other unit on both ends).
- You can monitor the status of the call on the OLED screen:
 - **CALLING**.
 - **CONNECTING** (depending on the communication interface and the network status, this may be a status of extremely brief duration).

- **SYNCHRONIZING** (depending on the communication interface and the network status, this may be a status of extremely brief duration).
- **CONNECTED.**
- Verify that the **SYNC** LED beneath the **CALL** button is lighted in green to indicate that the communication has been successfully established.
- Press the **ON AIR** button associated with that communications channel to enable audio to be transmitted over that interface in full duplex mode.
- Once the connection has been established with the remote end, confirm the presence of transmitted and received audio by checking the VU meters on the Phoenix STRATOS front panel.
 - If necessary, adjust the levels in **AUDIO SETTINGS→CODEC CHANNEL1 SETTINGS** (or **CODEC CHANNEL2 SETTINGS**) menu or in **ANALOG AUDIO SETTINGS** menu, according to the desired signal type.

5.5.1.1. Finalizing an IP communication in RTP mode.

- To finalize the communication, simply press the **CALL** button associated with the communications channel currently being used on the front panel of the unit. If, as recommended, “**SmartRTP**” is activated in both codecs involved, there is no need to repeat the hang up process in the remote end.

5.5.2. Establishing an IP communication in PROXY SIP mode.

- Ensure that the equipment is powered up.
- Check that the RJ45 cable is correctly connected and secured.
- Check that the amber indicator lamp in the communications module is blinking regularly.
- Verify the status of the communications interface. Ensure that no error message appears in the MAIN STATUS menu screen for this communications module.
- Establish the appropriate audio configuration (analog / digital input mode, gains, routing of inputs to the codec, suitable encoding, analog output modes).
- Set the VU meters at the most suitable adjustment point and check the input audio.
- Configure the SIP server in the **SYSTEM→IP PROVIDERS** menu. From the list, select one that you know is already configured, or one of the empty ones, and fill in the fields: enter the server address (either its IP or URL) and the physical network interface. Note that, if you enter the URL of the SIP server and not its IP address, you will have to configure and have connectivity with a DNS server (in **SYSTEM→SETTINGS→ETHERNET SETTINGS**).
- Go into the configuration of interface and select the “**PROXY SIP**” mode. Then press the “**ADVANCED**” context key. There you’ll be able to select, in “**PROVIDER**”, the SIP server from the list that you configured in the previous point, in “**USER**” and “**AUTHENTICATION**” the data from the SIP account to register yourself in the server (username, display name and password) and, in “**AUDIO**”, the physical interface for RTP audio (which may or may not be the same as the one used to communicate with the SIP server). Select the NAT resolver working mode (“**NAT TRAVERSAL**”) depending on the connection of the unit.

NOTE: We recommend you to follow the application notes 0-A or 0-C (depending on the connection of the unit).
- Assign the IP logic interface to the channel.
- Press the **CALL** button associated with the communications channel that is going to use the IP interface as the connection base; this action will open the connection configuration menu. Basically, it is a question at this point of ensuring that all the fields are correct, and that they correspond to what you have configured previously:
 - Check that the assigned IP interface is the desired one.
 - Select the encoding profile to be used in the communication. There are several established profiles, each of which contains one or more particular encodings, in order of preference. More profiles can be edited and added. The called unit will accept the call with the first encoding on the list it supports (whether or not it is in the profile that is established at that time).

- Select the working mode and the reception buffer size (see section 4.4 in order to select the optimal buffer configuration depending on your application).
- You have the possibility of using the backup option, with its corresponding configuration submenu in **BACK UP**:
 - Activate or not the backup option.
 - Select the interface to use as a backup.
 - Select the encoding mode. Make sure that the mode you select is compatible with the communications interface selected.
 - Define the minimum time interval for the connection to be considered down and the communication interface switching to be performed.
 - Select the number of retries to be made before considering the communication down.
 - Define the minimum time interval for the connection to be considered down if synchronization is not performed, and for the communication interface switching to be performed.
 - Define the destination toward which the backup connection is to be launched. Depending on the communication interface you have selected as a backup, this destination will be defined by an ISDN subscriber number, a SIP identifier (URI) or an IP address.
- Enter the IP identifier (URI) of the contact to be called, either manually or from the internal phone book in **CALL BOOK**. Enter the IP identifier in the following format: <unit_name>@<SIP_server_domain> (for instance, "phstr_xxx@sip.aeq.es"); when the unit you're calling to is registered in the same SIP server as the unit that's calling, it's enough just to use <name_unit>. **VERY IMPORTANT:** <unit_name> must be no longer than 19 characters.
- Press the **CALL** button on the front panel again, or select the **DIAL** option on screen, to make the call.
If you select the number to call from the phone book, you can perform the same action by using the contextual button associated with the **DIAL** option.
- You can monitor the status of the call on the OLED screen:
 - **CALLING.**
 - **CONNECTING** (depending on the communication interface and the network status, this may be a status of extremely brief duration).
 - **SYNCHRONIZING** (depending on the communication interface and the network status, this may be a status of extremely brief duration).
 - **CONNECTED.**



Detail of Phoenix STRATOS calling

- Verify that the **SYNC** LED beneath the CALL button is lighted in green to indicate that the communication has been successfully established.
- Press the **ON AIR** button associated with that communications channel to enable audio to be transmitted over that interface in full duplex mode.
- Once the connection has been established with the remote end, confirm the presence of transmitted and received audio by checking the VUmeters on the front panel.
 - If necessary, adjust the levels in **AUDIO SETTINGS**→**CODEC CHANNEL1 SETTINGS** (or **CODEC CHANNEL2 SETTINGS**) menu or in **ANALOG AUDIO SETTINGS** menu, according to the desired signal type.
- If you register the unit in the SIP server but there is no audio communication, please check the NAT TRAVERSAL configuration (see section 3.1.5.2.4).

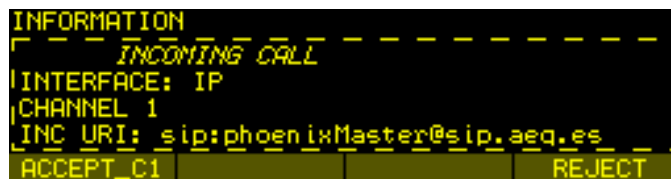
5.5.2.1. Finalizing an IP communication in PROXY SIP mode.

- To finalize the communication, simply press the **CALL** button associated with the communications channel currently being used on the front panel of the unit.

5.5.2.2. Receiving and accepting an IP communication in PROXY SIP mode.

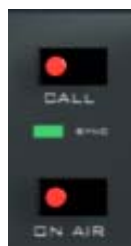
If the IP interface of the codec is suitably configured and the automatic answering mode is OFF, when a call is received:

- The unit will emit an audible notification signal. This can be disabled in **MENU SYSTEM → 8-SETTINGS → BUZZER.**
- The red **CALL** button light corresponding to the communications channel that receives the call will blink simultaneously to notify you.
- The caller's information will appear on the OLED screen:
 - The communications interface through which the call is being received.
 - The communications channel through which the call is being received. The channels and interfaces are associated in **MENU SYSTEM → 6-CHANNELS → Chn1 or Chn2 → INTERFACE.**
 - The calling unit identifier (URI).



Detail of the Phoenix STRATOS receiving a call

- To accept the call, press the **contextual button ACCEPT_C1** (when the call is received through communications channel 1) or **ACCEPT_C2** (when the call is received through communications channel 2).
You can also accept the call by pressing the **CALL** button associated with the communications channel that is receiving the incoming call.
- You can monitor the status of the call on the OLED screen:
 - **CONNECTING** (depending on the communication interface, this may be an extremely brief status).
 - **SYNCHRONIZING** (depending on the communication interface, this may be an extremely brief status).
 - **CONNECTED.**
- Verify that the **SYNC** LED beneath the **CALL** button is lighted in green to indicate that the communication has been successfully established.
- Press the **ON AIR** button associated with that communications channel to enable audio to be transmitted over that interface in full duplex mode.



- Once the connection has been established with the remote end, confirm the presence of transmitted and received audio by checking the VU meters on the Phoenix STRATOS front panel.
 - If necessary, adjust the levels in **AUDIO SETTINGS→CODEC CHANNEL1 SETTINGS** (or **CODEC CHANNEL2 SETTINGS**) menu or in **ANALOG AUDIO SETTINGS** menu, according to the desired signal type.

The automatic answering mode can be activated or deactivated in the **MENU SYSTEM → 6-CHANNELS → Chn1 or Chn2 → AUTOANSWER.**

5.5.3. Establishing an IP communication in DIRECT SIP mode.

- Ensure that the equipment is powered up.
- Check that the RJ45 cable is correctly connected and secured.
- Check that the amber indicator lamp in the communications module is blinking regularly.
- Verify the status of the communications interface. Ensure that no error message appears in the MAIN STATUS menu screen for this communications module.
- Establish the appropriate audio configuration (analog / digital input mode, gains, routing of inputs to the codec, suitable encoding, analog output modes).
- Set the VU meters at the most suitable adjustment point and check the input audio.
- Go into the interface configuration and select the “DIRECT SIP” mode. Then press the “ADVANCED” context key. There you will be able to select the unit name in “USER” and the physical interface for audio in “AUDIO”. Select the NAT resolver working mode (“NAT TRAVERSAL”) depending on the connection of the unit.

NOTE: We recommend you to follow the application notes 0-A or 0-C (depending on the connection of the unit).

- Assign the IP logic interface to the channel.
- Press the **CALL** button associated with the communications channel that is going to use the IP interface as the connection base; this action will open the connection configuration menu. Basically, it is a question at this point of ensuring that all the fields are correct, and that they correspond to what you have configured previously:
 - Check that the assigned IP interface is the desired one.
 - Select the encoding profile to be used in the communication. There are several established profiles, each of which contains one or more particular encodings, in order of preference. More profiles can be edited and added.
 - Select the working mode and the reception buffer size (see paragraph 4.4 of this manual in order to select the optimal buffer configuration depending on your application).
 - You have the possibility of using the backup option, with its corresponding configuration submenu in **BACK UP**:
 - Activate or not the backup option.
 - Select the interface to use as a backup.
 - Select the encoding mode. Make sure that the mode you select is compatible with the communications interface selected, especially as regards the associated bit rates.
 - Define the minimum time interval for the connection to be considered down and the communication interface switching to be performed.
 - Select the number of retries to be made before considering the communication down.
 - Define the minimum time interval for the connection to be considered down if synchronization is not performed, and for the communication interface switching to be performed.
 - Define the destination toward which the backup connection is to be launched. Depending on the communication interface you have selected as a backup, this destination will be defined by an ISDN subscriber number, a SIP identifier (URI) or an IP address.
 - Enter the identifier of the contact to be called, either manually or from the internal phone book in **CALL BOOK**. Enter the IP identifier in the following format: <unit_name>@<unit_IP_address> (for instance, phstr_xxx@192.168.1.84). **VERY IMPORTANT:** <unit_name> must be no longer than 19 characters.
- Press the **CALL** button on the front panel again, or select the **DIAL** option on screen, to make the call.

If you select the number to call from the phone book, you can perform the same action by using the contextual button associated with the **DIAL** option.
- You can monitor the status of the call on the OLED screen:
 - **CALLING.**
 - **CONNECTING** (depending on the communication interface, this may be an extremely brief status).

- **SYNCHRONIZING** (depending on the communication interface, this may be an extremely brief status).
- **CONNECTED.**



Detail of Phoenix STRATOS calling

- Verify that the **SYNC** LED beneath the CALL button is lighted in green to indicate that the communication has been successfully established.
- Press the **ON AIR** button associated with that communications channel to enable audio to be transmitted over that interface in full duplex mode.
- Once the connection has been established with the remote end, confirm the presence of transmitted and received audio by checking the VU meters on the Phoenix STRATOS front panel.
 - If necessary, adjust the levels in **AUDIO SETTINGS** → **CODEC CHANNEL1 SETTINGS** (or **CODEC CHANNEL2 SETTINGS**) menu or in **ANALOG AUDIO SETTINGS** menu, according to the desired signal type.

5.5.3.1. Finalizing an IP communication in DIRECT SIP mode.

- To finalize the communication, simply press the **CALL** button associated with the communications channel currently being used on the front panel of the unit.

5.5.3.2. Receiving and accepting an IP communication in DIRECT SIP mode.

If the IP interface of the codec is suitably configured and the automatic answering mode is OFF, when a call is received:

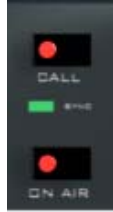
- The unit will emit an audible notification signal. This can be disabled in **MENU SYSTEM** → **8-SETTINGS** → **BUZZER**.
- The red **CALL** button light corresponding to the communications channel that receives the call will blink simultaneously to notify you.
- The caller's information will appear on the OLED screen:
 - The communications interface through which the call is being received.
 - The communications channel through which the call is being received. The channels and interfaces are associated in **MENU SYSTEM** → **6-CHANNELS** → **Chn1** or **Chn2** → **INTERFACE**.
 - The calling number identifier.



Detail of the Phoenix STRATOS receiving a call

- To accept the call, press the **contextual button ACCEPT_C1** (when the call is received through communications channel 1) or **ACCEPT_C2** (when the call is received through communications channel 2).
You can also accept the call by pressing the **CALL** button associated with the communications channel that is receiving the incoming call.
- You can monitor the status of the call on the OLED screen:
 - **CONNECTING** (depending on the communication interface, this may be an extremely brief status).
 - **SYNCHRONIZING** (depending on the communication interface, this may be an extremely brief status).

- **CONNECTED.**
- Verify that the **SYNC** LED beneath the **CALL** button is lighted in green to indicate that the communication has been successfully established.
- Press the **ON AIR** button associated with that communications channel to enable audio to be transmitted over that interface in full duplex mode.



- Once the connection has been established with the remote end, confirm the presence of transmitted and received audio by checking the VU meters on the Phoenix STRATOS front panel.
 - If necessary, adjust the levels in **AUDIO SETTINGS** → **CODEC CHANNEL1 SETTINGS** (or **CODEC CHANNEL2 SETTINGS**) menu or in **ANALOG AUDIO SETTINGS** menu, according to the desired signal type.

The automatic answering mode can be activated or deactivated in the **MENU SYSTEM** → **6-CHANNELS** → **Chn1** or **Chn2** → **AUTOANSWER**.

6. QUICKSTART USER'S GUIDE. REMOTE CONTROL.

To gain a complete grasp of the PHOENIX STRATOS, we **recommend** reading the previous chapters and "AEQ ControlPHOENIX" user's manual carefully. The paragraphs below describe the basic actions you will need to take for **remote** operation of the equipment by means of "AEQ ControlPHOENIX" application. **IMPORTANT NOTE:** For that purpose, the equipment must be previously configured to accept remote control connections (see **section 3.1.7** of this manual).

6.1. Equipment connections.

6.1.1. Power supply.

The two 3-pin IEC power feed connectors are located on the back panel of the codec. Connecting one is enough.

6.1.2. Communications interfaces.

Depending on the type of communication, a certain type of connection will be needed:

- **IP connection:** Connect Ethernet cable (CAT5 or higher) terminated in an RJ45 10/100 BT connector to the connector installed for this purpose in the rear panel of the unit. The selected cable will be the direct 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 2.2.7 of this manual.
- **ISDN connection:** Connect standard cable terminated in an RJ11 or RJ45 connector to one of the two connectors made available for this purpose in the rear section of the ISDN module, according to the protocol and standard to be used (see section 2.2.8).
- **V.35 connection:** Connect standard cable terminated in a DB25 connector to the connector installed for this purpose in the rear section of the unit (see section 2.2.5).

6.2. Powering up the codec.

- Once the codec has been connected to the mains, after about 15 seconds, the OLED screen in the display will turn on, showing the AEQ logo and the name of the unit.



- After a few seconds, the MAIN STATUS menu screen will be displayed. Check to see that there is no ERROR message (the screen has dimmed lighting, press any key to see it properly).



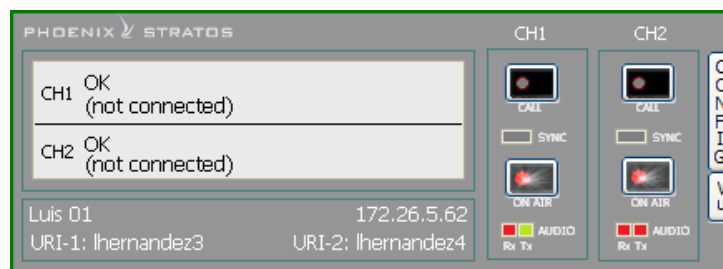
- Configure the unit to accept remote control (see **section 3.1.7** of this manual).
- The unit is ready to be used.

6.3. Setting up a computer to control the unit.

Connect to the same network a computer with “AEQ ControlPHOENIX” software installed. Follow what’s indicated in chapter 3 of the application’s user manual.

Check that your PHOENIX STRATOS is automatically discovered after launching the application, according to chapter 4.1 of the software user’s manual. Accept the unit and, if you don’t find it, please check that the network parameters of both the unit and the computer belong to the same network having in mind the default IP address of the unit (**192.168.1.87**, GWAY=DNS=192.168.1.100).

The codec is ready to be controlled when the individual codec control window appears, showing “OK” label in CH1 and CH2 status areas and, below, the equipment’s given name (“Luis 01” in the example below).



IMPORTANT NOTE: If more new codecs will be controlled in the same network, you need to change their IP address one by one as you add them in order to avoid conflicts in the network, as they will also have the same default IP addresses. Go to “Configuration” → “Ethernet” to access the dialog that allows you to change the IP parameters of the unit.

6.4. Audio.

Section 2.2 of this manual describes in detail the physical connections at the back panel of the unit, but the procedure would be, in a nutshell, as follows:

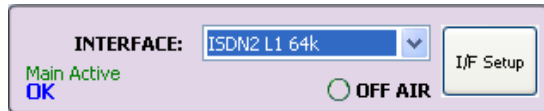
- Connect the line inputs in analog or digital format that you need, according to the description of the connectors installed in the PHOENIX STRATOS rear panel. XLR connectors for the analog inputs and DB15 connector for the digital inputs.
- From General Configuration screen, select, for each channel the input signal type to be used between “Analog” or “Digital”, and whether it is “Stereo”, you want only “Mono L”, “Mono R”, or “Mono L+R” mix.
- From General Configuration screen, select the working mode for the stereo analog output: “Stereo” “Mono L”, “Mono R” o “Mono L+R”.
- Connect the line outputs in analog or digital format that you need, according to the description of the connectors installed in the PHOENIX STRATOS rear panel. XLR connectors for the analog outputs and DB15 connector for the digital outputs.
- There is no need to define the output type because the PHOENIX STRATOS offers, by default, all the outputs duplicated in analog and digital format.

6.5. Establishing an ISDN communication.

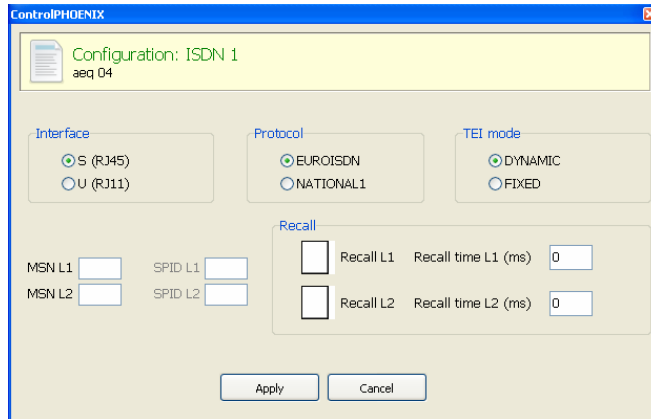
- Ensure that the equipment is powered up and controlled by the software.
- Establish the appropriate audio configuration (analog/digital input mode, gains, routing of inputs to the codec, suitable encoding, analog output mode).
- Check that there is incoming audio to the channel (CH1 or CH2) we are going to use to establish the communication: the “Tx” indicator in the individual codec control window

 , in the general configuration screen   and in the list view  will change to green .

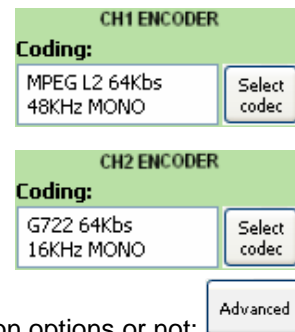
- Go to general configuration screen and configure "INTERFACE" as "ISDN1 L1+L2 128k", "ISDN1 L1 64k" or "ISDN1 L2 64k".



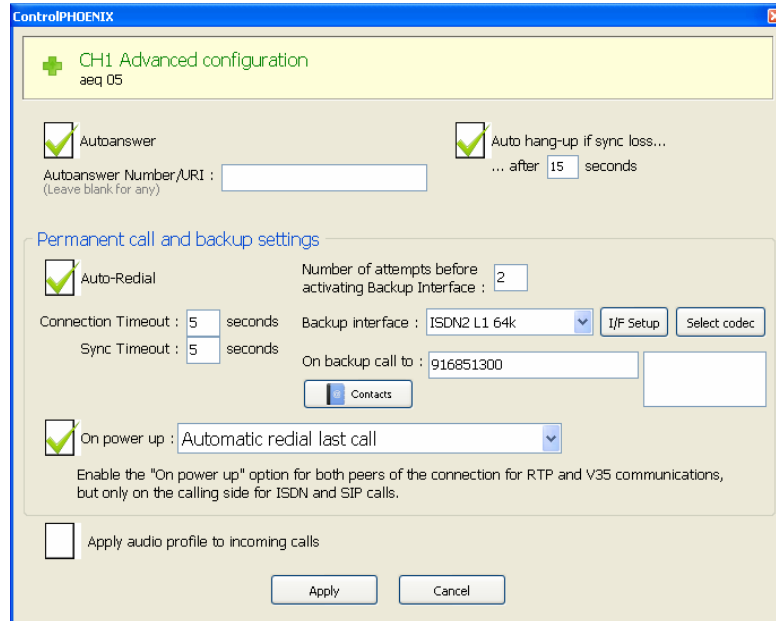
- Enter "I/F Setup" and configure the type of physical interface will be selected (S or U), the ISDN protocol (EuroISDN or National 1) and whether the TEI is dynamic or fixed. We will also indicate the MSN numbers (SPID if protocol is National 1), whether we want automatic recall or not and its timing.



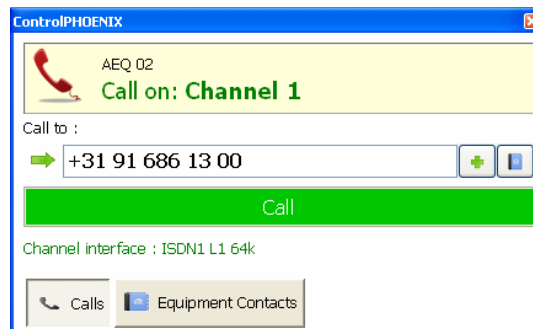
- Return to the general configuration screen, check that the selected coding mode in the green "ENCODER" area corresponding to that channel (CH1 or CH2) is correct, or otherwise click on "Select codec" to change it. Make sure that the mode you select is compatible with the communications interface selected, especially as regards the associated bit rates.



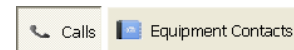
- Decide whether you will use the advanced automatic connection options or not:
 - "Autoanswer": Automatic call answering for all incoming calls, or only those corresponding to a predefined caller.
 - "Auto hang-up": Automatic hang-up whenever audio packets are missed for a given time. This option is not available for ISDN and V35.
 - "Permanent call and backup settings": Allows you to activate automatic redial and configure a backup interface and define the time and number of retries before switching, the number or unit to call to and the used encoding. You can select no backup interface ("NONE") and activate the same way the "Auto-Redial" check box when you simply want a call to be retried in case of connection falls. The "On power up" check box and the associated drop-down menu allow you to activate the automatic redial when equipment starts up. This option should be activate for both peers of the connection for RTP or V35 communications and only on the calling side for ISDN or SIP calls.
 - "Apply audio profile to incoming calls": allows you to filter the SIP calls depending on the encoding profile of the receiver. This option is not applied for ISDN communications.



- Return to the individual codec control window and click on **“CALL”** button corresponding to that channel, showing then the call screen:

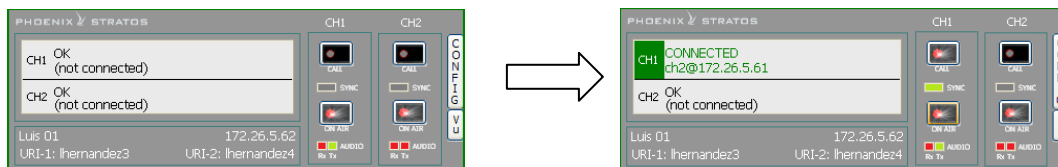


Enter the number of the contact to be called, either manually or getting it from the buttons:



By clicking here, the last called numbers (“Call”) and the numbers in the “Equipment Contacts” book are shown respectively, but only those with formats compatible with channel and communication type.

- Press the green **“Call”** button on the screen, to make the call.
- You can observe the changes of status from OK to CONNECTED and synchronized when calling from channel 1:



- Verify that the **“SYNC”** LED beneath the **“CALL”** button is lighted in green to indicate that the communication has been successfully established.
- Press the **“ON AIR”** button to allow the transmission of full-duplex audio for that channel.
- Send audio from one equipment to another verifying that the **“Tx”** and **“Rx”** audio presence indicators change to green.



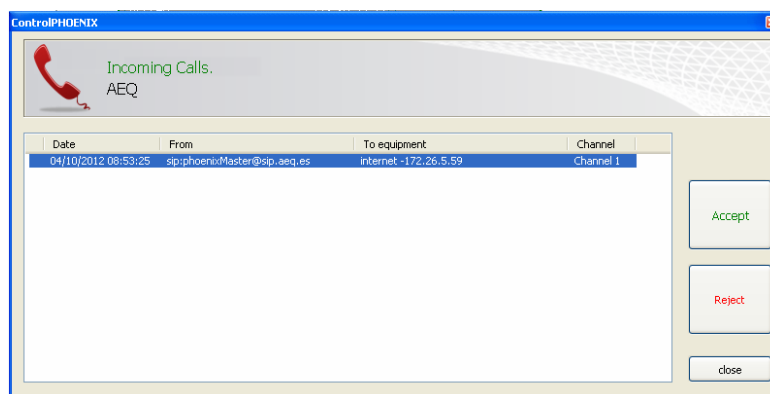
6.5.1. Finalizing an ISDN communication.

- In order to hang call, just click on “**CALL**” button of the connected channel in one of both codecs. A confirmation message will appear and the call will be disconnected after acceptance.

6.5.2. Receiving and accepting an ISDN communication.

If the ISDN interface is correctly configured and automatic answer mode is OFF, when you receive a call:

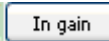
- The unit and the application will provide acoustic warning. This can be disabled (for the unit) at “**Configuration**” → “**Miscellaneous**” → “**Buzzer and test**”.
- The “**CALL**” button red LED of the called channel at the individual codec control window in the remote control software corresponding to the unit that is receiving a call will blink at the same time to warn the user.
- In addition, an incoming call window will appear showing the number of the caller unit:



- The call will be accepted by clicking on the individual codec control window “**CALL**” button of the called channel or, alternatively, on the “**Accept**” button in the incoming calls window.
- The screen will show the status of the call:
 - **CONNECTING** (depending on the network status this status can last for an extremely short time).
 - **SYNCHRONIZING** (depending on the network status this status can last for an extremely short time).
 - **CONNECTED.**
- Verify that the “**SYNC**” LED beneath the “**CALL**” button corresponding to that channel is lighted in green to indicate that the communication has been successfully established.
- Press the “**ON AIR**” button to allow the transmission of full-duplex audio for that channel.
- Send audio from one equipment to another verifying that the “Tx” and “Rx” audio presence indicators change to green.
- If necessary, adjust the output levels in the “**att**” box at OUTPUTS block in “**Configuration**” → “**General**” screen. You can also make an additional adjustment from the windows you access by pressing “**In gain**” button in “**ENCODER**” green window and “**Out gain**” button in “**DECODER**” green window.

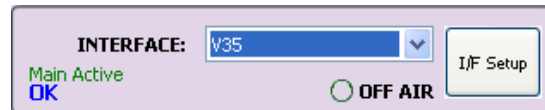
6.6. Establishing a V35 communication.

- Ensure that the equipment is powered up and controlled by the software.
- Establish the appropriate audio configuration (analog/digital input mode, gains, routing of inputs to the codec, suitable encoding, analog output mode).
- Check that there is incoming audio to the channel (CH1 or CH2) we are going to use to establish the communication: the “Tx” indicator in the individual codec control window

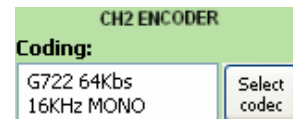


, in the general configuration screen will change to green .

- Go to general configuration screen and configure “INTERFACE” as “V35”.



- Check that the selected coding mode in the green “ENCODER” area corresponding to that channel (CH1 or CH2) is correct, or otherwise click on “Select codec” to change it. Make sure that the mode you select is compatible with the communications interface selected, especially as regards the associated bit rates.

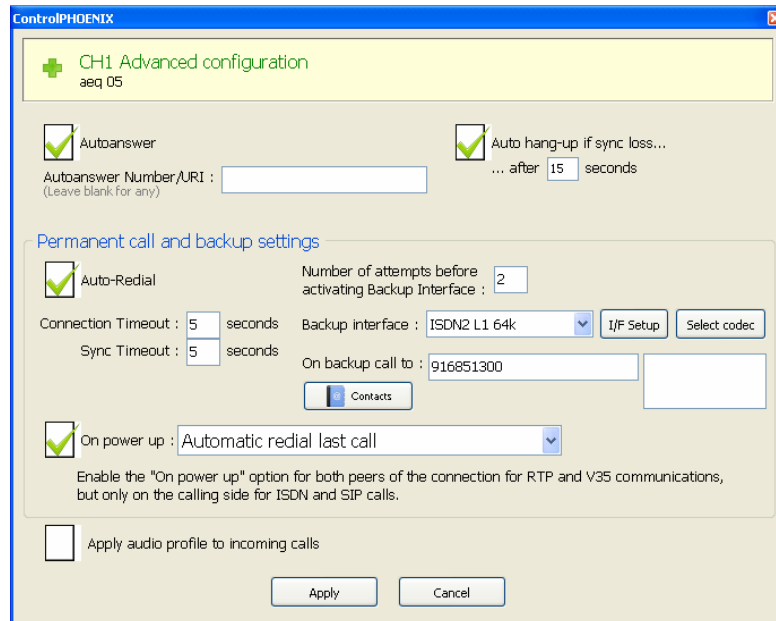


- Decide whether you will use the advanced automatic connection options or not:
 - “Autoanswer”: Automatic call answering for all incoming calls, or only those corresponding to a predefined caller.
 - “Auto hang-up”: Automatic hang-up whenever audio packets are missed for a given time. This option is not available for ISDN and V35.
 - “Permanent call and backup settings”: Allows you to activate automatic redial and configure a backup interface and define the time and number of retries before switching, the number or unit to call to and the used encoding. You can select no backup interface (“NONE”) and activate the same way the “Auto-Redial” check box when you simply want a call to be retried in case of connection falls.

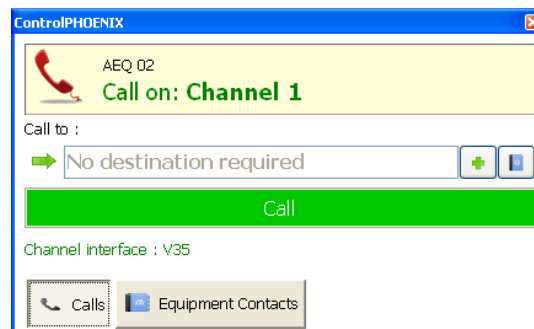
When selected mode is RTP or V35 and “Auto-Redial” option is active, when synchronism is lost the call will not be disconnected: it saves time of recovery and allows to keep the call connected when there is no return (for instance, in multiunicast links).

The “On power up” check box and the associated drop-down menu allow you to activate the automatic redial when equipment starts up. This option should be activate for both peers of the connection for RTP or V35 communications and only on the calling side for ISDN or SIP calls.
 - “Apply audio profile to incoming calls”: allows you to filter the SIP calls depending on the encoding profile of the receiver. This option is not applied for V35 communications.



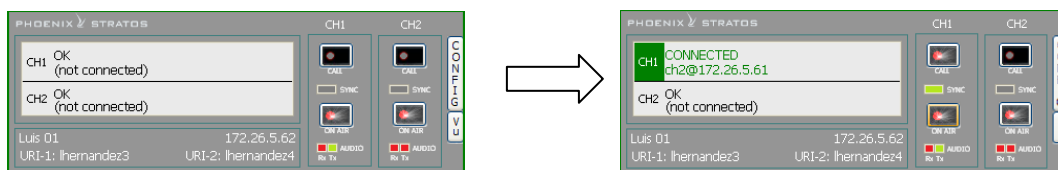


- Return to the individual codec control window and click on **“CALL”** button corresponding to that channel, showing then the call screen:



In the case of a V35 communication, no destination is required in that screen.

- Press the green **“Call”** button on the screen, to make the call.
- You can observe the changes of status from OK to CONNECTED and synchronized when calling from channel 1:



- Verify that the **“SYNC”** LED beneath the **“CALL”** button is lighted in green to indicate that the communication has been successfully established.
- Press the **“ON AIR”** button to allow the transmission of full-duplex audio for that channel.
- Send audio from one equipment to another verifying that the **“Tx”** and **“Rx”** audio presence indicators change to green.



6.6.1. Finalizing a V35 communication.

- To finalize the communication, simply press the “CALL” button in the individual codec control window and then confirm. Repeat the process in the unit at the other end.




6.7. Establishing an IP communication.

As explained in previous chapters, several operating modes are available, depending on the protocol used for the communication initiation.

In order to ease the task, AEQ has developed the proprietary “SmartRTP” protocol, that allows for the establishment of a communication without having to dial from both ends and without worrying about the coding algorithms matching, as the calling end will provide all the necessary signaling to its counterpart, so it knows what IP address and port to send the return audio to. This functionality is, in some sense, similar to the one provided by DIRECT SIP, but without renouncing to RTP inherent simplicity (all other modes are based on RTP, indeed) and without the need for additional special control ports.

6.7.1. Establishing an IP communication in RTP mode using SmartRTP.

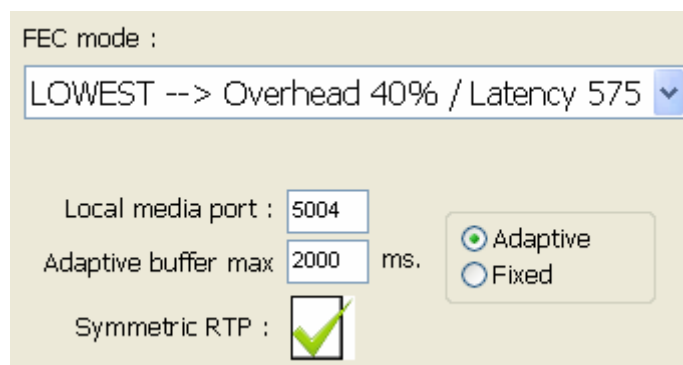
- Ensure that the equipment is powered up and controlled by the software.
- Establish the appropriate audio configuration (analog/digital input mode, gains, routing of inputs to the codec, suitable encoding, analog output mode).
- Check that there is incoming audio to the channel (CH1 or CH2) that we are going to use to establish the communication: the “Tx” indicator in the individual codec control window

 , in the general configuration screen  and in the list view  will change to green .

- Go to general configuration screen and configure “INTERFACE” as “RTP raw”.

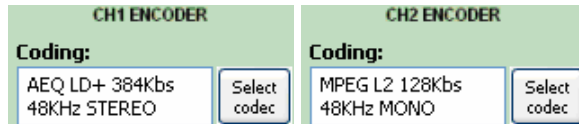


- Enter “I/F Setup” and select the “Local media port”: local IP port through which the RTP audio is received. The remote unit, when calling, will send audio to that port (see section 4.2.3).



The same screen allows you to configure the type and size of the receiving buffer and FEC parameters as a function of the IP network quality so we have the shortest delay while audio cuts are minimized or eliminated in poor quality networks (see paragraph 4.4 of this manual in order to select the optimal buffer configuration depending on your application).

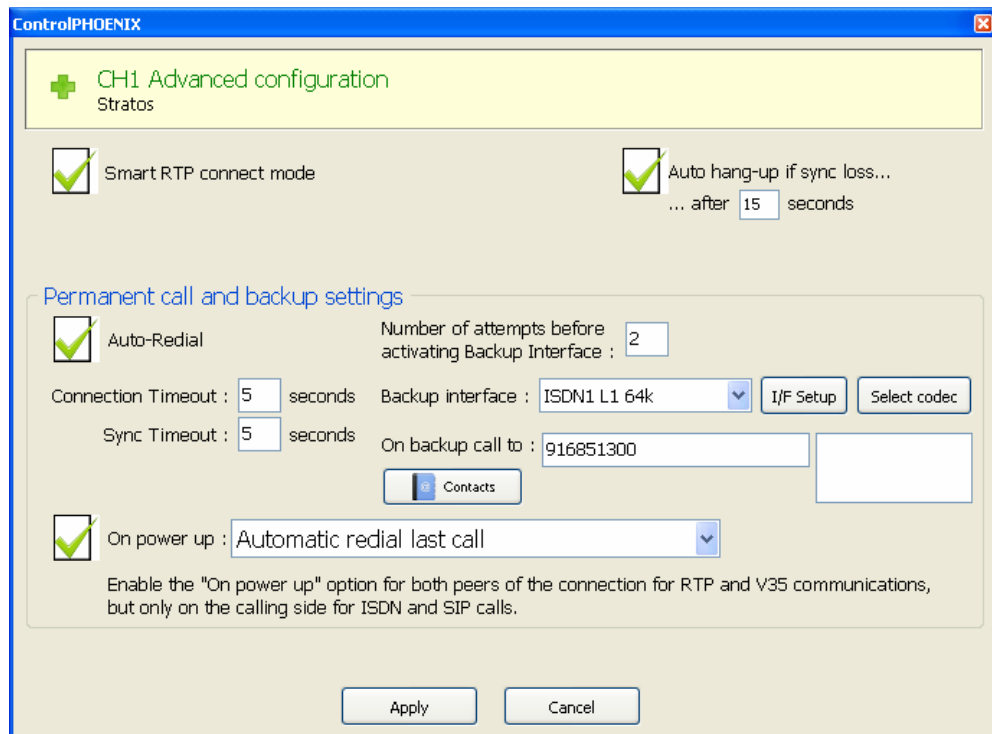
- Return to the general configuration screen, check that the selected coding algorithm in the green “ENCODER” area corresponding to that channel (CH1 or CH2) is correct, or otherwise click on “Select codec” to change it.



- Decide whether you will use the advanced automatic connection options or not: Advanced
 - “**SmartRTP connect mode**”: Activate this option.
 - “Auto hang-up”: Automatic hang-up whenever audio packets are missed for a given time.
 - “Permanent call and backup settings”: Allows you to activate automatic redial and configure a backup interface and define the time and number of retries before switching, the number or unit to call to and the used encoding. You can select no backup interface (“NONE”) and activate the same way the “Auto-Redial” check box when you simply want a call to be retried in case of connection falls. When “**SmartRTP**” mode is active, it is recommended that this option is enabled in the calling end only.

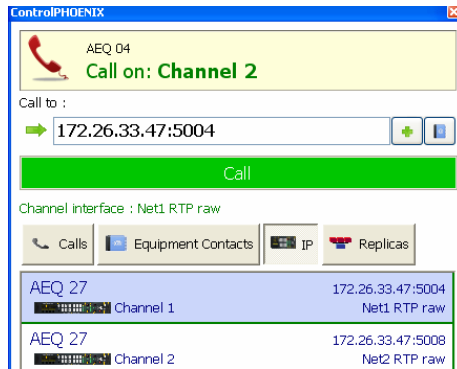
When selected mode is RTP without “*SmartRTP*” or V35 and “Auto-Redial” option is active, when synchronism is lost the call will not be disconnected: it saves time of recovery and allows to keep the call connected when there is no return (for instance, in multiunicast links).

The “On power up” check box and the associated drop-down menu allow you to activate the automatic redial when equipment starts up. This option should be activate for both peers of the connection for RTP without “*SmartRTP*” or V35 communications and only on the calling side for ISDN, SIP or RTP with “*SmartRTP*” calls.



- Return to the individual codec control window and click on “**CALL**” button of the desired channel, showing then the call screen:





Enter the IP address of the remote unit, either manually or getting it from the buttons:



By clicking here, the last URI “Calls”, the URIs in the “Contacts” book or the available “IP” addresses are shown respectively, but only those with formats compatible with channel and communication type, in this case only IP addresses followed by “:<port>”.




- Press the green “Call” button on the screen, to make the call.
- Repeat the audio and mode configuration in the other end. As “SmartRTP” is enabled, the other codec will automatically connect as soon as it starts receiving audio traffic, so it won’t be necessary to set the coding mode, dial IP addresses or ports, etc. nor even accept the incoming call. Only when “SmartRTP” is not active, you’ll have to repeat the calling process at the other end (because RTP is connection-oriented, you also need to connect to the IP of the other unit on both ends).
- You can monitor the status of the call on the screen:
 - **CALLING.**
 - **CONNECTING** (depending on the communication interface and the network status, this may be a status of extremely brief duration).
 - **SYNCHRONIZING** (depending on the communication interface and the network status, this may be a status of extremely brief duration).
 - **CONNECTED.**
- Verify that the “SYNC” LED beneath the “CALL” button is lighted in green to indicate that the communication has been successfully established.
- Press the “ON AIR” button associated with that communications channel to enable audio to be transmitted over that interface in full duplex mode.
- Once the connection has been established with the remote end, confirm the presence of transmitted and received audio by checking the audio presence indicators “Tx” and “Rx”.
- If necessary, adjust the output levels in the “att” box at OUTPUTS block in “Configuration” → “General” screen. You can also make an additional adjustment from the windows you access by pressing “In gain” button in “ENCODER” green window and “Out gain” button in “DECODER” green window.

NOTE: In order to make calls to **multiple destinations**, please consult section 4.2.3.1 of this manual.

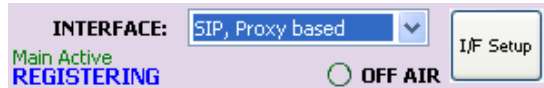
6.7.1.1. Finalizing an IP communication in RTP mode.

- To finalize the communication, simply press the “CALL” button in the individual codec control window and then confirm. If, as recommended, “SmartRTP” is activated in both codecs involved, there is no need to repeat the hang up process in the remote end.

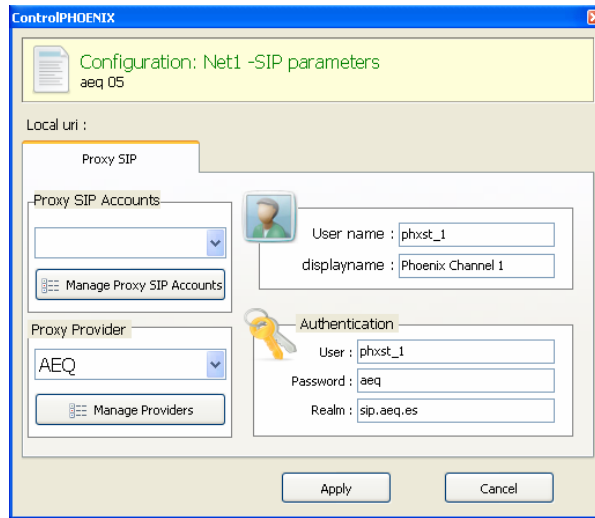
6.7.2. Establishing an IP communication in PROXY SIP mode.

- Ensure that the equipment is powered up and controlled by the software.
- Establish the appropriate audio configuration (analog/digital input mode, gains, routing of inputs to the codec, suitable encoding, analog output mode).
- Check that there is incoming audio to the channel (CH1 or CH2) that we are going to use to establish the communication: the “Tx” indicator in the individual codec control window  , in the general configuration screen  and in the list view  will change to green .

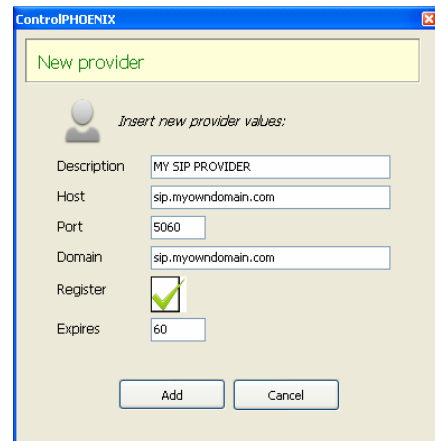
- Go to general configuration screen and configure “INTERFACE” as “SIP, Proxy based”.



- Enter “I/F Setup” and click on “SIP Parameters”.



Check the SIP server configuration (“Proxy Provider”). Select one that is already configured from the list, for example the default “AEQ”, or otherwise go to “Manage Providers”, create a “New provider” and fill in the following fields: name, port, address of the server (either its IP or URL), SIP interface and, if necessary, check the “Register” field. In this case, you also need to re-write the “Authentication” data in the channel’s SIP parameters so they match those in the new server, as the ones by default are only for “AEQ”. Note that, in case you specify the SIP Server by its URL instead of its IP address, the DNS Server must be correctly configured and reachable (at “Configuration” → “Ethernet”).

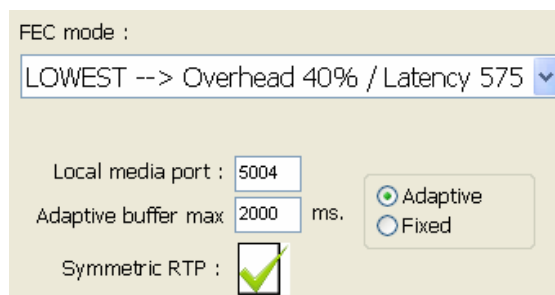


You can also select a SIP account from a previously created and stored list by means of the drop-down menu “Proxy SIP Accounts”. See section 5.1.7.1 of “AEQ ControlPHOENIX” user’s manual.

- Select the working mode to traverse NAT devices (“**NAT Traversal**”) that is more adequate for the network the unit is connected to.

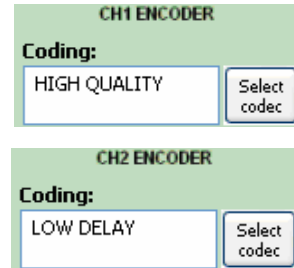
NOTE: It is recommended that you follow Application Notes 0-A or 0-C, according to the type of equipment’s connection.

At “I/F Setup” fill in the “Local media port” (where the unit expects to receive RTP audio traffic at). If you enable Symmetric RTP mode, the unit will send audio to the same port where it is receiving it from. This is sometimes useful to overcome NAT routers.

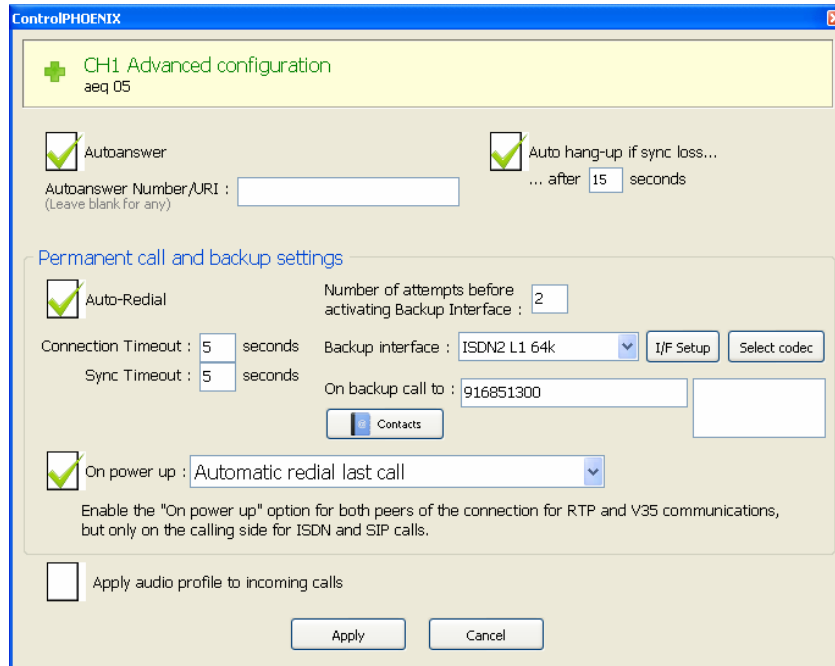


The same screen allows you to configure the type and size of the receiving buffer and FEC parameters as a function of the IP network quality so we have the shortest delay while audio cuts are minimized or eliminated in poor quality networks (see paragraph 4.4 of this manual in order to select the optimal buffer configuration depending on your application).

- Return to the general configuration screen, check that the selected coding algorithm in the green “ENCODER” area corresponding to that channel (CH1 or CH2) is correct, or otherwise click on “Select codec” to change it. There are several pre-defined profiles containing several particular algorithms each one, with preference ordering. They can be edited and more profiles can be added. The called unit will accept the call using the first coding algorithm that it supports from the list (independently of the profile selected in that unit at that time).

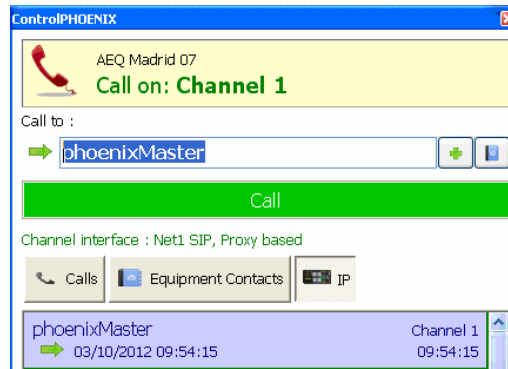


- Decide whether you will use the advanced automatic connection options or not:
 - “Autoanswer”: Automatic call answering for all incoming calls, or only those corresponding to a predefined caller.
 - “Auto hang-up”: Automatic hang-up whenever audio packets are missed for a given time.
 - “Permanent call and backup settings”: Allows you to activate automatic redial and configure a backup interface and define the time and number of retries before switching, the number or unit to call to and the used encoding. You can select no backup interface (“NONE”) and activate the same way the “Auto-Redial” check box when you simply want a call to be retried in case of connection falls. The “On power up” check box and the associated drop-down menu allow you to activate the automatic redial when equipment starts up. This option should be activate for both peers of the connection for RTP or V35 communications and only on the calling side for ISDN or SIP calls.
 - “Apply audio profile to incoming calls”: allows you to filter the SIP calls depending on the encoding profile of the receiver.

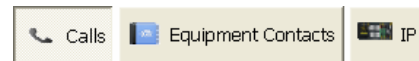


- Return to the individual codec control window and click on “CALL” button corresponding to that channel, showing then the call screen:





Enter the IP address of the remote unit, either manually or getting it from the buttons:

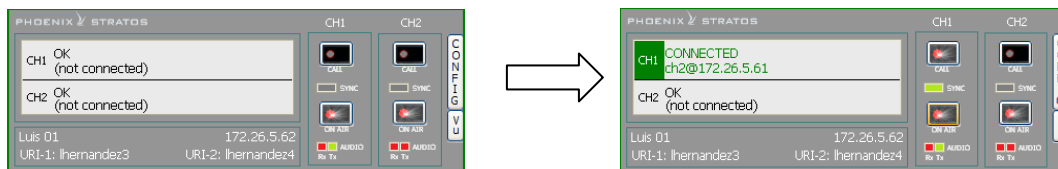


By clicking here, the last URI “Calls”, the URIs in the “Equipment Contacts” book or the available “IP” addresses are shown respectively, but only those with formats compatible with channel and communication type.

- It is mandatory that the called unit URI is specified in any of the following formats, adequate for Proxy SIP communications:
 - **<equipment’s_name>**, for example: “phstr_231” or “phoenixMaster”
 - **<equipment’s_name>@<SIP_realm>**, for example: “phstr_231@sip.aeq.es” or phoenixMaster@sip.aeq.es”
 - **<equipment’s_name>@<SIP_server_IP>**, for example: “phstr_231@232.168.1.2” or “phoenixMaster@232.168.1.2” (where 232.168.1.2 is AEQ SIP Server (“sip.aeq.es”) IP address)
 - **<equipment’s_name>@<SIP_server>:<Port>**, when SIP port is not 5060 (the one used by default in SIP SERVER mode). For example: “phstr_231@sip.aeq.es:5061”

VERY IMPORTANT: <equipment’s_name> must not be longer than 19 characters.

- Press the green “Call” button on the screen, to make the call.
- You can observe the changes of status from OK to CONNECTED and synchronized:



- Verify that the “**SYNC**” LED beneath the “**CALL**” button corresponding to that channel is lighted in green to indicate that the communication has been successfully established.
- Press the “**ON AIR**” button associated with that communications channel to enable audio to be transmitted over that interface in full duplex mode.
- Send audio from one equipment to another verifying that the “Tx” and “Rx” audio presence indicators change to green.
- If the unit is registering in SIP server and the call is being made but no audio comes through, please check “**NAT TRAVERSAL**” configuration (see section 4.3).



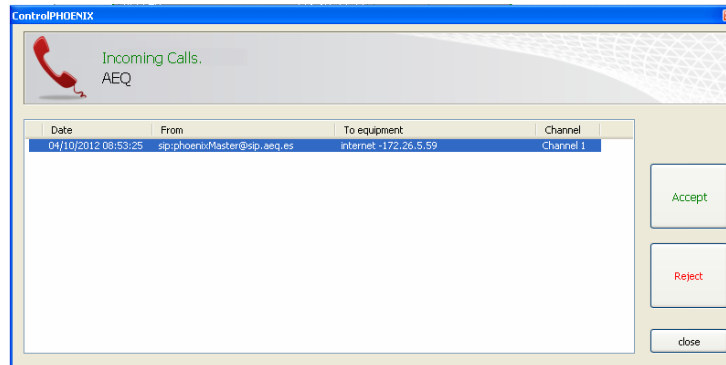
6.7.2.1. Finalizing an IP communication in PROXY SIP mode.

- In order to hang call, just click on “**CALL**” button corresponding to that channel in one of both codecs’ individual control window. A confirmation message will appear and the call will be disconnected after acceptance.

6.7.2.2. Receiving and accepting an IP communication in PROXY SIP mode.

If the IP interface is correctly configured and automatic answer mode is OFF, when you receive a call:

- The unit and the application will provide acoustic warning. This can be disabled (for the unit) at **“Configuration” → “Miscellaneous” → “Buzzer and test”**.
- The **“CALL”** button red LED corresponding to that channel at the individual codec control window in the remote control software corresponding to the unit that is receiving a call will blink at the same time to warn the user.
- In addition, an incoming call window will appear showing the URI identifier of the caller unit:



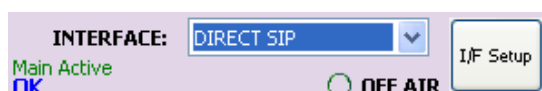
- The call will be accepted by clicking on the individual codec control window **“CALL”** button corresponding to that channel or alternatively on the **“Accept”** button in the incoming calls window.
- The screen will show the status of the call:
 - **CONNECTING** (depending on the network status this status can last for an extremely short time).
 - **SYNCHRONIZING** (depending on the network status this status can last for an extremely short time).
 - **CONNECTED.**
- Verify that the **“SYNC”** LED beneath the **“CALL”** button corresponding to that channel is lighted in green to indicate that the communication has been successfully established.
- Press the **“ON AIR”** button to allow the transmission of full-duplex audio for that channel.
- Send audio from one equipment to another verifying that the **“Tx”** and **“Rx”** audio presence indicators change to green.
- If necessary, adjust the output levels in the **“att”** box at OUTPUTS block in **“Configuration” → “General”** screen. You can also make an additional adjustment from the windows you access by pressing **“In gain”** button in **“ENCODER”** green window and **“Out gain”** button in **“DECODER”** green window.

6.7.3. Establishing an IP communication in DIRECT SIP mode.

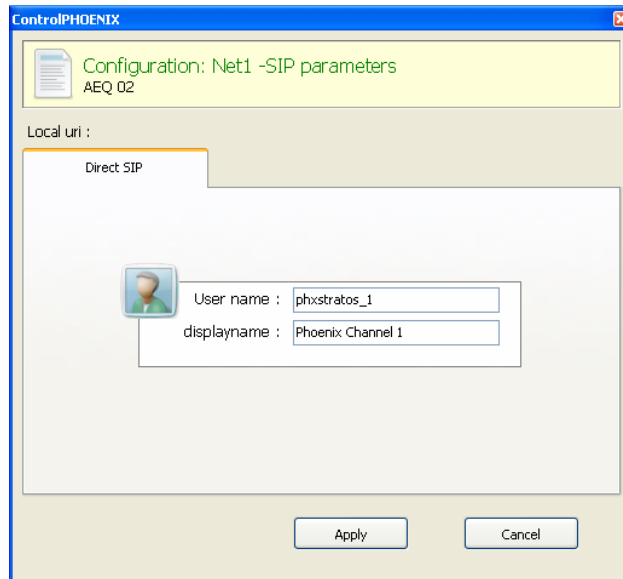
- Ensure that the equipment is powered up and controlled by the software.
- Establish the appropriate audio configuration (analog/digital input mode, gains, routing of inputs to the codec, suitable encoding, analog output mode).
- Check that there is incoming audio to the channel (CH1 or CH2) we are going to use to establish the communication: the **“Tx”** indicator in the individual codec control window



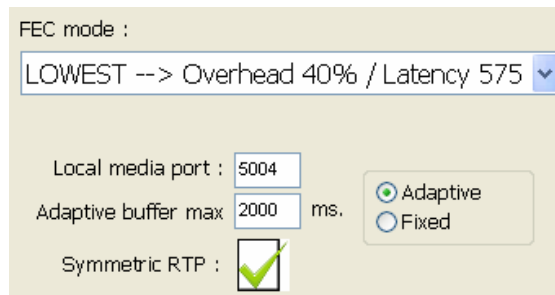
- Go to general configuration screen and configure **“INTERFACE”** as **“DIRECT SIP”**.



- Enter “I/F Setup” and click on “SIP Parameters”. Check that “User Name” and “Display name” are configured. User name and IP address constitute the equipment’s required connection information.

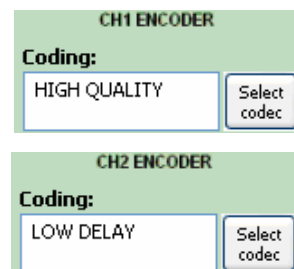


- Select the working mode to traverse NAT devices (“**NAT Traversal**”) that is more adequate for the network the unit is connected to.
NOTE: It is recommended that you follow Application Notes 0-A or 0-C, according to the type of equipment’s connection.
- At “I/F Setup” fill in the “Local media port” (where the unit expects to receive RTP audio traffic at). If you enable Symmetric RTP mode, the unit will send audio to the same port where it is receiving it from. This is sometimes useful to overcome NAT routers.

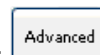


The same screen allows you to configure the type and size of the receiving buffer and FEC parameters as a function of the IP network quality so we have the shortest delay while audio cuts are minimized or eliminated in poor quality networks (see paragraph 4.4 of this manual in order to select the optimal buffer configuration depending on your application).

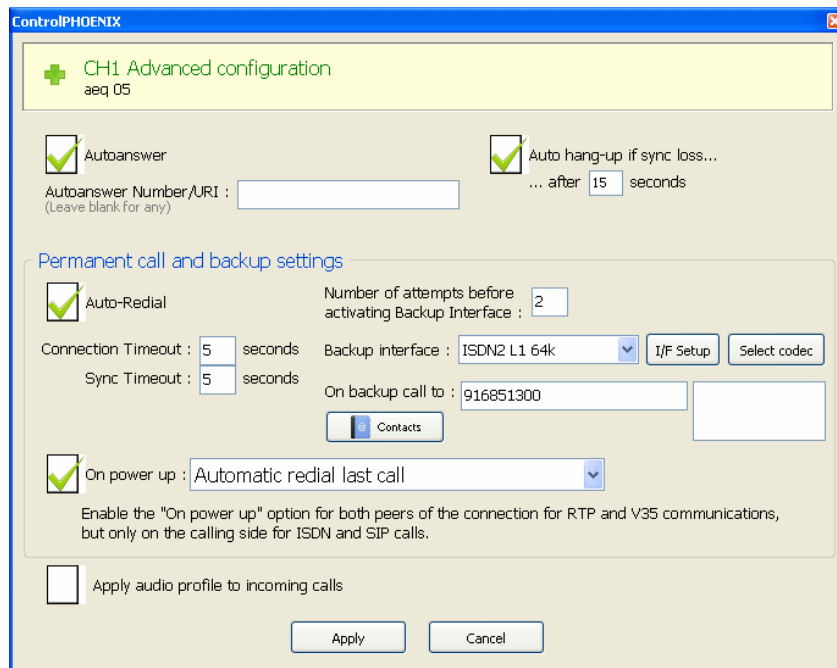
- Return to the general configuration screen, check that the selected coding algorithm in the green “ENCODER” area corresponding to that channel (CH1 or CH2) is correct, or otherwise click on “Select codec” to change it. There are several pre-defined profiles containing several particular algorithms each one, with preference ordering. They can be edited and more profiles can be added. The called unit will accept the call using the first coding algorithm that it supports from the list (independently of the profile selected in that unit at that time).



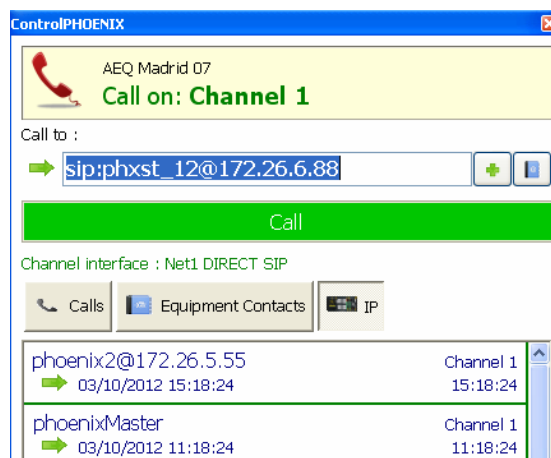
- Decide whether you will use the advanced automatic connection options or not:



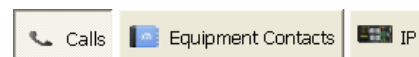
- “Autoanswer”: Automatic call answering for all incoming calls, or only those corresponding to a predefined caller.
- “Auto hang-up”: Automatic hang-up whenever audio packets are missed for a given time.
- “Permanent call and backup settings”: Allows you to activate automatic redial and configure a backup interface and define the time and number of retries before switching, the number or unit to call to and the used encoding. You can select no backup interface (“NONE”) and activate the same way the “Auto-Redial” check box when you simply want a call to be retried in case of connection falls.
The “On power up” check box and the associated drop-down menu allow you to activate the automatic redial when equipment starts up. This option should be activate for both peers of the connection for RTP or V35 communications and only on the calling side for ISDN or SIP calls.
- “Apply audio profile to incoming calls”: allows you to filter the SIP calls depending on the encoding profile of the receiver.



- Return to the individual codec control window and click on “CALL” button corresponding to that channel, showing then the call screen:



Enter the IP address of the remote unit, either manually or getting it from the buttons:



By clicking here, the last URI “Calls”, the URIs in the “Equipment Contacts” book or the available “IP” addresses are shown respectively, but only those with formats compatible with channel and communication type.

- It is mandatory that the called unit URI is specified in any of the following formats, adequate for Direct SIP communications:

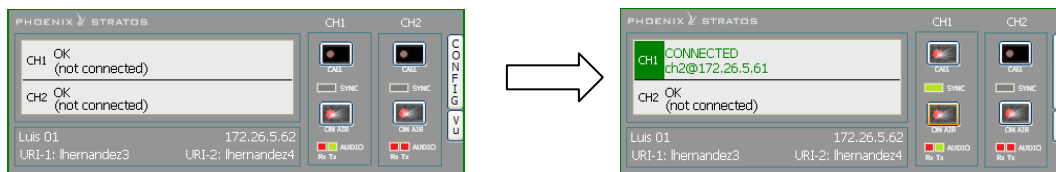
“<equipment’s_name>@<equipment’s_IP_address>” (for example: phxst_231@172.26.5.57”).

If the **SIP port** of the other end is not 5060 (standard SIP port), the identifier must include the port in use:

For example: “phxst_231@172.25.32.11:5061”.

VERY IMPORTANT: <equipment’s_name> must not be longer than 19 characters

- Press the green “**Call**” button on the screen, to make the call.
- You can observe the changes of status from OK to CONNECTED and synchronized when calling from channel 1:



- Verify that the “**SYNC**” LED beneath the “**CALL**” button is lighted in green to indicate that the communication has been successfully established.
- Press the “**ON AIR**” button to allow the transmission of full-duplex audio for that channel.
- Send audio from one equipment to another verifying that the “Tx” and “Rx” audio presence indicators change to green.



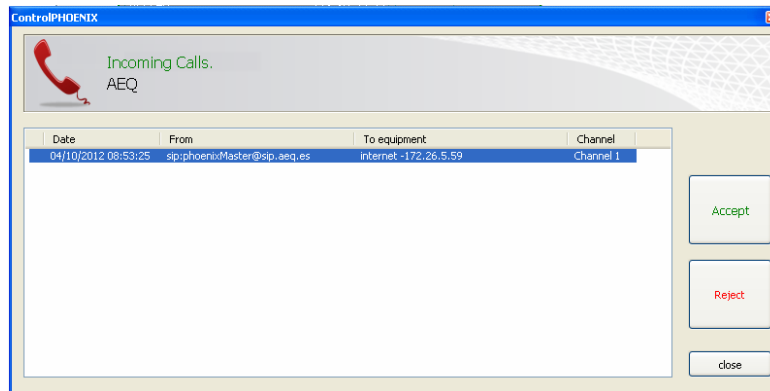
6.7.3.1. Finalizing an IP communication in DIRECT SIP mode.

- In order to hang call, just click on “**CALL**” button of the connected channel in one of both codecs. A confirmation message will appear and the call will be disconnected after acceptance.

6.7.3.2. Receiving and accepting an IP communication in DIRECT SIP mode.

If the IP interface is correctly configured and automatic answer mode is OFF, when you receive a call:

- The unit and the application will provide acoustic warning. This can be disabled (for the unit) at “**Configuration**” → “**Miscellaneous**” → “**Buzzer and test**”.
- The “**CALL**” button red LED of the called channel at the individual codec control window in the remote control software corresponding to the unit that is receiving a call will blink at the same time to warn the user.
- In addition, an incoming call window will appear showing the URI identifier of the caller unit:



- The call will be accepted by clicking on the individual codec control window “**CALL**” button of the called channel or, alternatively, on the “**Accept**” button in the incoming calls window.
- The screen will show the status of the call:
 - **CONNECTING** (depending on the network status this status can last for an extremely short time).
 - **SYNCHRONIZING** (depending on the network status this status can last for an extremely short time).
 - **CONNECTED.**
- Verify that the “**SYNC**” LED beneath the “**CALL**” button corresponding to that channel is lighted in green to indicate that the communication has been successfully established.
- Press the “**ON AIR**” button to allow the transmission of full-duplex audio for that channel.
- Send audio from one equipment to another verifying that the “Tx” and “Rx” audio presence indicators change to green.
- If necessary, adjust the output levels in the “**att**” box at OUTPUTS block in “Configuration” → ”General” screen. You can also make an additional adjustment from the windows you access by pressing “In gain” button in “**ENCODER**” green window and “Out gain” button in “**DECODER**” green window.

7. CONTROL TERMINAL OVER WEB BROWSER.

The PHOENIX STRATOS includes a WebServer that enables you to execute numerous functions remotely over the Ethernet interface included in the back panel of the unit; with the aid of a standard web browser (compatibility is guaranteed with Internet Explorer running on Microsoft Windows operating system).

7.1. Upgrading the equipment firmware.

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

Because the upgrading process must be handled with caution, we recommend having it done by an authorized distributor, or under the instructions of the AEQ Technical Assistance Service. If questions or problems arise, consult the Technical Assistance Service via electronic mail (sat@aeq.es).

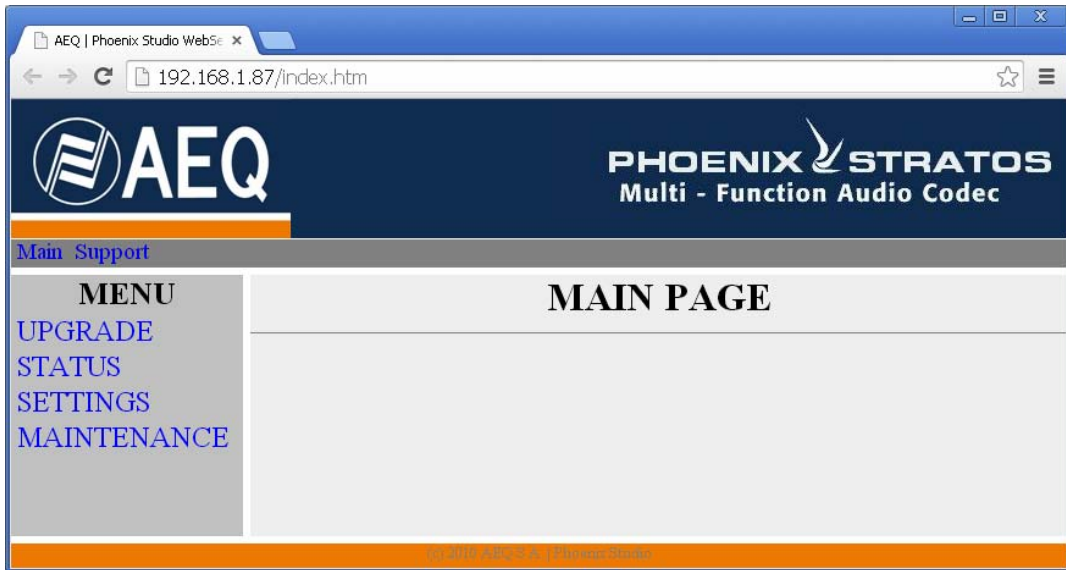
IMPORTANT NOTE: If the **MICRO (CPU)** of the equipment is upgraded, part of its configuration may be modified, specifically the part having to do with the MAC addresses, URIs, the SIP Proxy Server Users and the associated Passwords. It is important to make a note of these codec parameters before upgrading in order to reconfigure them afterwards.

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

IMPORTANT NOTE: the recommended order for upgrading is: MICRO (CPU), DSP, FPGA followed by the ISDN communication module . The process is iterative.

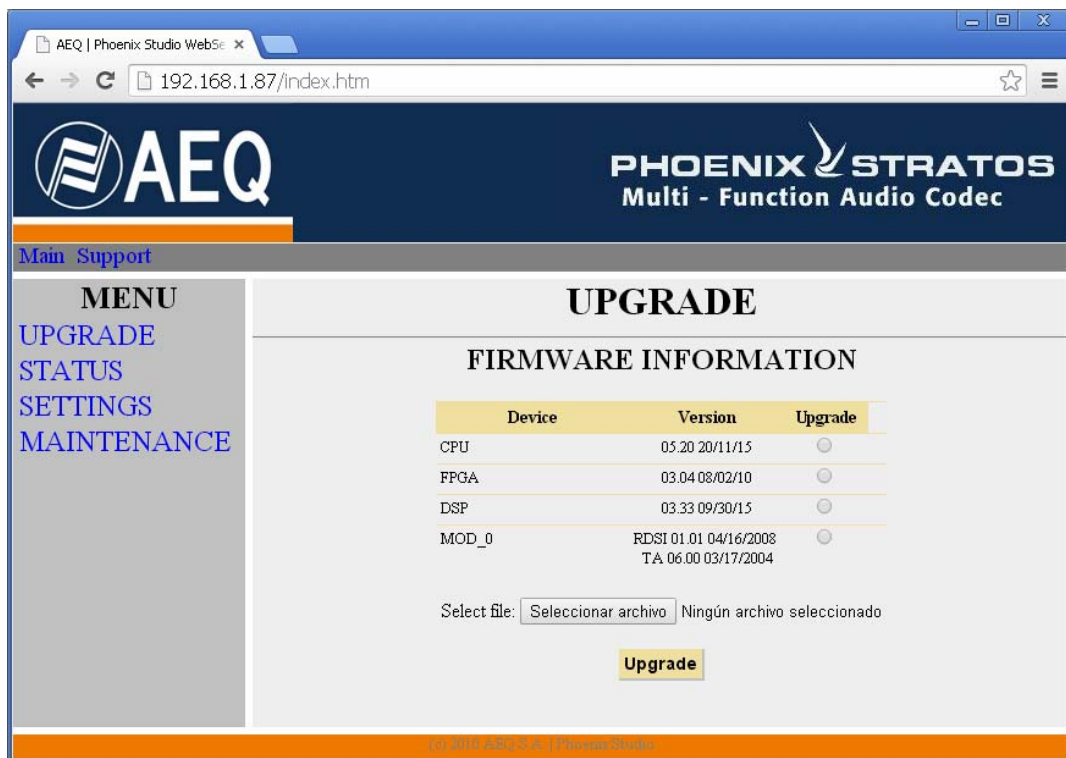
To upgrade the firmware, you must follow the steps detailed below:

1. From **MAIN STATUS → SYSTEM → 8-SETTINGS → ETHERNET SETTINGS → ETHERNET CONFIG**, check the IP address associated with the Ethernet interface.
2. Power down the PHOENIX STRATOS.
3. Connect PHOENIX STRATOS to the PC from which you are going to perform the upgrading process using a crossed cable.
4. Power up the PHOENIX STRATOS.
5. Open the Internet Explorer web browser and, in the address bar, enter **HTTP://<IP address obtained in point 1>**. Press ENTER and the main screen will be displayed.



Main screen detail

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



Firmware upgrading screen detail

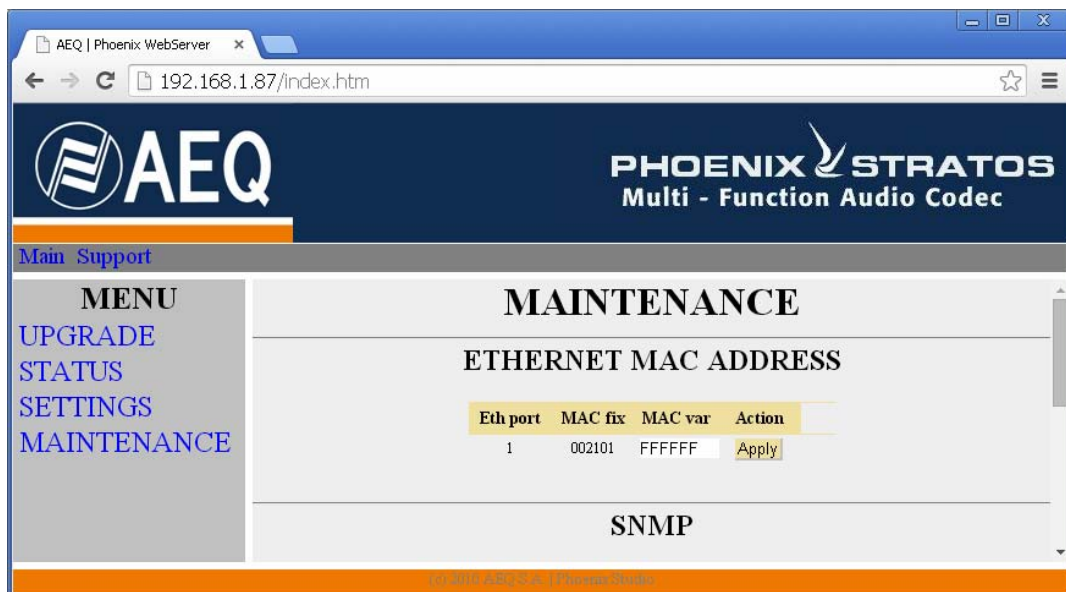
8. Check to see whether the versions displayed are the same as the firmware that is currently in effect. If they do not match, upgrade the firmware as indicated below.
9. Select the module you want to upgrade in "Upgrade" column. **NOTE:** Each upgrading file is specifically designed to upgrade a specific module within the unit: CPU, DSP, FPGA or ISDN communication module (MOD_0).
10. In "Select file" enter the access route to the upgrading file containing the new firmware version, using the "Seleccionar archivo" button.
11. Press the "Upgrade" button in the lower part of the screen.

12. Wait for on-screen confirmation that the operation has been successfully completed.
13. In the Internet browser, go to the **UPGRADE** section and ensure that all the firmware versions installed in your codec are now the correct ones.
14. Power the unit down.

7.2. Configuring the MAC address associated with the Ethernet interface.

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

1. From **MAIN STATUS** → **SYSTEM** → **8-SETTINGS** → **ETHERNET SETTINGS** → **ETHERNET CONFIG**, check the IP address associated with the Ethernet interface.
2. Power down the PHOENIX STRATOS.
3. Connect PHOENIX STRATOS to the PC from which you are going to perform the upgrading process using a crossed cable.
4. Power up the PHOENIX STRATOS.
5. Open the Internet Explorer web browser and, in the address bar, enter HTTP://<IP address obtained in point 1>. Press ENTER and the main screen will be displayed. Selecting the **MAINTENANCE** option will enable you to modify the MAC address of the Ethernet interface of the unit.



MAC change screen detail

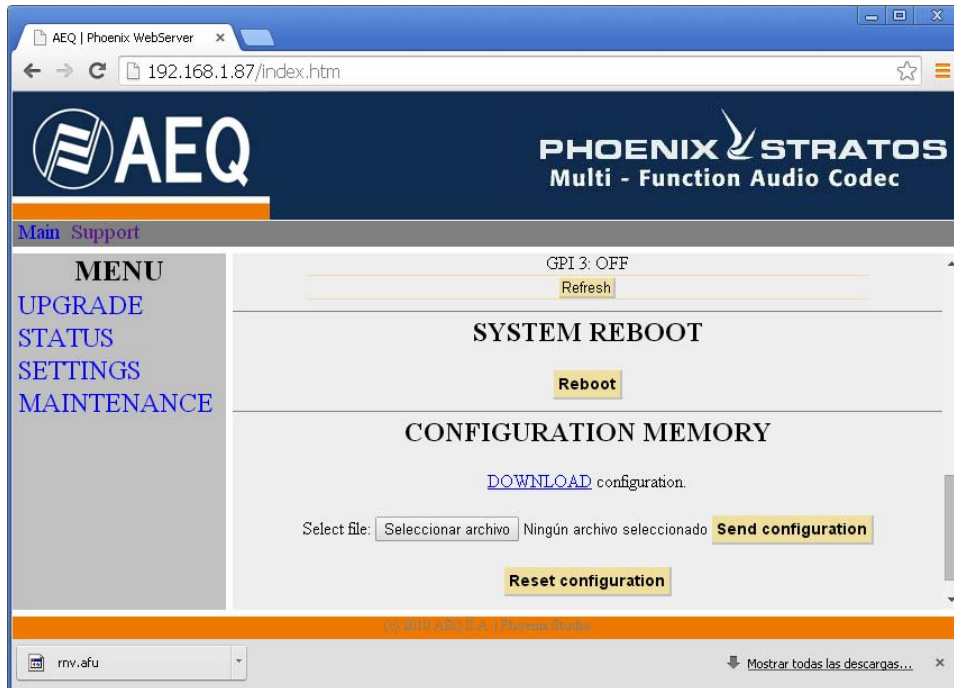
6. Modify the value in the MAC field associated with the desired Ethernet interface.
7. Press the **Apply** button.
8. Wait for on-screen confirmation that the operation has been successfully completed.
9. In the Internet browser, go to the **MAINTENANCE** section and ensure that the MAC addresses are now the correct ones.
10. Power the unit down.

7.3. Technical Assistance Service and on-line manuals.

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

7.4. Saving and loading configurations.

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

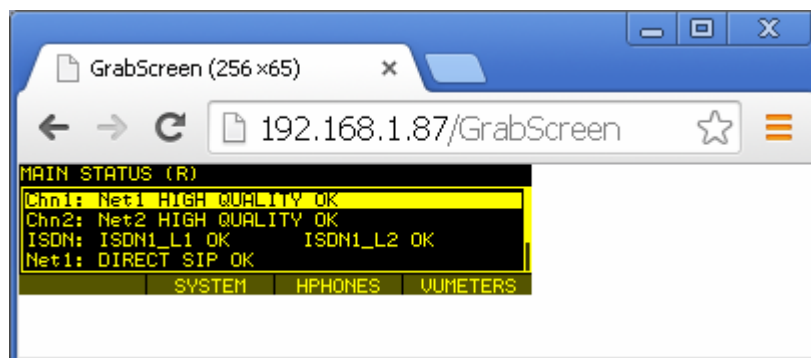


7.5. Screen capture.

The unit allows you to capture the screen displayed in the multifunction screen at any moment in order to save it into a PC in BMP image format. This function may be useful in order to prepare documents or report problems or doubts to SAT.

In order to capture a screen, open the web browser and write the following text in address bar:
http://<unit_IP_address>/GrabScreen

A new screen will appear showing the image. Place the pointer on the image, press the right mouse button and select “Save Image as” (the message may vary depending on the web browser).



Screen capture detail

7.6. GPOs and GPIs configuration.

You can find the **GPOs** and **GPIs** configuration area inside **MAINTENANCE** section, in the middle of the screen. There you can monitor and manually set up General Purpose Inputs and Outputs, respectively.

In case of **GPOs**, you can control its value, forcing a logical value:

GPOs	
GPO	State
0	<input checked="" type="checkbox"/>
1	<input type="checkbox"/>
2	<input type="checkbox"/>
3	<input type="checkbox"/>

[Apply](#)

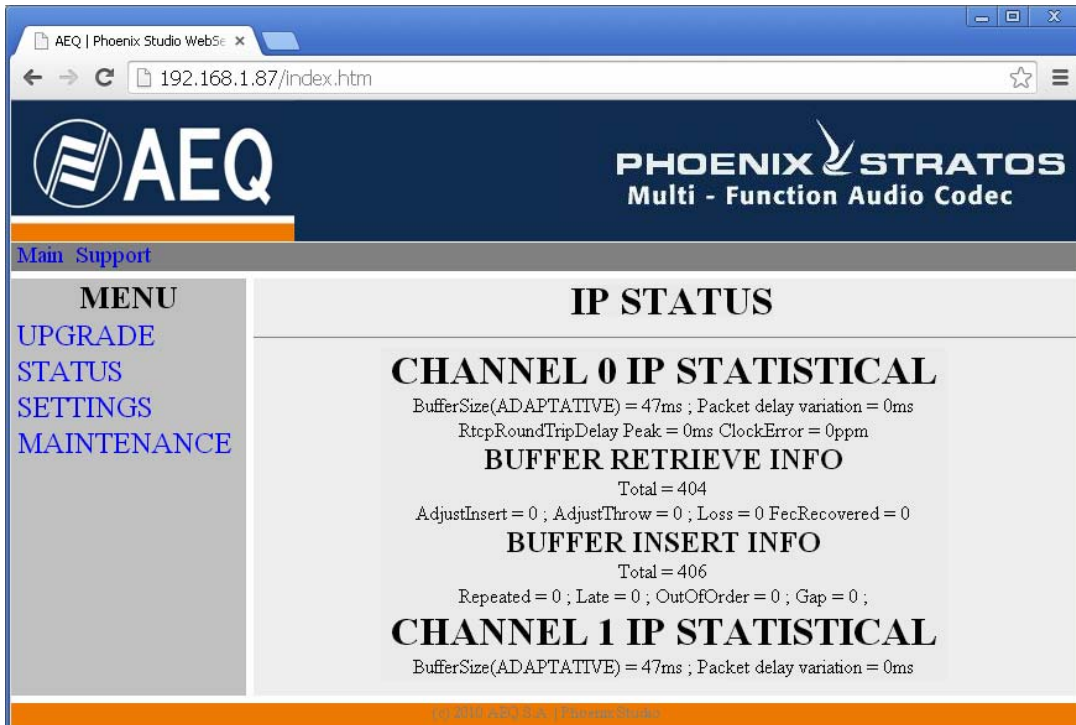
In case of **GPIs**, their status will be readed, showing this information in the associated screen:

GPIs	
GPI 0:	OFF
GPI 1:	OFF
GPI 2:	OFF
GPI 3:	OFF

[Refresh](#)

7.7. Status menu.

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



IP Status screen detail

7.8. SNMP.

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

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

7.8.1. Getting the Phoenix STRATOS MIB file.

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

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

For more information, please consult section 4.10 of this manual and section 6.5.1 of “AEQ ControlPHOENIX” application manual.

7.9. Remotely rebooting the equipment.

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

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

8. TECHNICAL SPECIFICATIONS*

Audio Inputs and Outputs	
2 ST Analog line input	4 x XLR female. 9Kohm. Electronic balancing. Line level.
2 ST Digital line input	DB15 connector. Dual AES/EBU interface for independent inputs (accepts any sampling rate between 24 and 192KHz), configurable to a single dual input by channel.
2 ST Analog line output	4 x XLR male. Output impedance < 100 ohm. Electronic balancing. Line level.
2 ST Digital line output	Two AES/EBU audio outputs with SRC through DB15 connector. Output sampling frequency of 48KHz. Variable to any frequency between 24 and 48KHz by means of an external synchronism source (when source's frequency is higher than 48KHz, digital output will also work synchronized, but at a submultiple frequency, that is when input is working at 96 or 192KHz output will work at 48KHz but synchronized with input).
Monitor headphones output	Headphones 1 x 1/4" Jack ST with volume control. Independent assignment of left and right headphones.
Synchronism	Each AES / EBU output can be synchronized through its input AES / EBU in the corresponding DB15 connector
Audio Characteristics	
Nominal output level	+0dBV
Maximum output level	+20 dB over nominal
Maximum input level	+20dBV (+22dBu), THD<0.5%
Digital audio interface	AES/EBU
Output impedance	<100 ohm
Distortion	<0.03% (depends on selected encoding algorithm)
Dynamic range	> 105dB for digital signals
Crosstalk	< -70dB
Bit rate	64 / 128Kbps
Mode	Mono, Stereo, Dual, JStereo, MS Stereo
Frequency response	+/-0.5dB @ 20Hz-20KHz
Sampling rate	
Analog I/O A/D D/A converter	24 bit Sigma-Delta 48 kHz max.
ISDN sampling rate	24 kHz, 32kHz, 48 kHz Configurable via software
Communication interfaces	
IP interface	RJ-45 Ethernet Port
ISDN interface	ISDN module with National-1 and EuroISDN interfaces.
Satellite	An external satellite telephone can be connected to the IP interface. (see application note AN2)
3G modem	An external 3G terminal can be connected to the IP interface. (see application note AN5)
Backup	Selectable by menu
Compression Algorithm	
IP	OPUS, G.711, G.722, MPEG Layer 2, AAC, PCM... See ANNEXE A
ISDN	G.711, G.722, MPEG Layer 2, AAC (by special order), hybrid modes... See ANNEXE A

Data and control interface	
10 Base / 100 base T LAN	RJ-45 connector
GPI: General Purpose Input	4 x 2.7-50 V optocouplers
GPO: General Purpose Output	4 x Open collectors, Max. 50V 50 mA.
General characteristics	
Weight	3.5 Kgs
Dimensions	W x D x H (mm): 486 x 280 x 44
Display	1 x OLED 256x64 pixels. 16 intensity levels. 2 x stereo multicolor VU meters

* 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.

9. A.E.Q. WARRANTY.

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

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

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

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

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

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

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

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

5.- The present guarantee will not be applicable in the following situations:
Improper use or Contrary use of the product as per the User or Instruction Manual; violent manipulation; exhibition to humidity or extreme thermal or environmental conditions or sudden changes of such conditions; electrical discharges or lightning; oxidation; modifications or not authorized connections; repairs or non-authorized disassembly of the product; spill of liquids or chemical products.

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

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

ANNEXE A: General characteristics of encoding modes.

OPUS:

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

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

G.711:

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

Bandwidth: 3.5 KHz

For further information on this subject, consult:

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

G.722:

ITU encoding standard, based on ADPCM algorithms, for processing audio signals in the human voice frequency band, through the compression of digital audio samples obtained at 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: 7 KHz.

For further information on this subject, consult:

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

MPEG LAYER II:

Well known, widely accepted encoding mode that is used when the delay is not important, since MPEG modes always have a greater delay than G.722 modes. There are 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.

AEQ LD+:

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

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.

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

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

Linear audio without any compress process.

For further information on this subject, consult

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

IMPORTANT NOTE: AAC-LC and AAC-LD modes are optional, they are not available in the basic firmware version the unit is supplied with. For further information, please consult the AEQ Sales Department or authorized distributors.

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

List of available coding algorithms in Phoenix STRATOS depending on the selected communication interface. Check availability for other algorithms.

CODEC	RATE (Kbps)	Fs (KHz)	Mode	IP (RTP)	IP (SIP)	ISDN	V35
OPUS	12	48	VOICE RB	YES	YES	-	-
	20	48	VOICE	YES	YES	-	-
	32	48	MUSIC MONO RB	YES	YES	-	-
	64	48	MUSIC MONO	YES	YES	-	-
	64	48	MUSIC STEREO RB	YES	YES	-	-
	192	48	MUSIC STEREO HQ	YES	YES	-	-
G.711 A-law	64	8	MONO	YES	YES	YES	-
G.711 μ -law	64	8	MONO	YES	YES	YES	-
G.722	64	16	MONO	YES	YES	YES	YES
AEQ-LD+	64	16	MONO	YES	YES	-	-
	128	16	STEREO	YES	YES	-	-
	128	32	MONO	YES	YES	YES	YES
	192	48	MONO	YES	YES	-	-
	256	32	STEREO	YES	YES	-	YES
MPEG-1 Layer II	64	32 / 48	MONO	YES	YES	YES	YES
	128	32 / 48	M / ST / JST	YES	YES	YES	YES
	192	32 / 48	M / ST / JST / D	YES	YES	-	-
	256	32 / 48	ST	YES	YES	-	YES
	384	48	STEREO	YES	YES	-	-
MPEG-2 Layer II	64	16 / 24	MONO	YES	YES	YES	YES
	128	24	MONO	YES	YES	-	-
AAC-LC	32	24 / 32 / 48	MONO	YES	YES	-	-
	64	24 / 32 / 48	M / ST / MST	YES	YES	YES	YES
	96	24 / 32 / 48	M / ST / MST	YES	YES	-	-
	128	24 / 32 / 48	M / ST	YES	YES	YES	YES
	192	24 / 32 / 48	M / ST	YES	YES	-	-
	256	48	M / ST	YES	YES	-	YES
AAC-LD	32	48	MONO	YES	YES	-	-
	64	48	M / ST / MST	YES	YES	YES	YES
	96	48	M / ST / MST	YES	YES	-	-
	128	48	M / ST	YES	YES	YES	YES
	192	48	M / ST	YES	YES	-	-
	256	48	M / ST	YES	YES	-	YES
PCM	<768	32	M / ST	YES	YES	-	-
	<1152	48	M / ST	YES	YES	-	-
	<1024	32	M / ST	YES	YES	-	-
	<1536	48	M / ST	YES	YES	-	-
	<1280	32	M / ST	YES	YES	-	-
	<2M	48	M / ST	YES	YES	-	-
	<1536	32	M / ST	YES	YES	-	-
<2.4M	48	M / ST	YES	YES	-	-	

IMPORTANT NOTE: AAC-LC and AAC-LD modes are optional, they are not available in the basic firmware version the unit is supplied with. For further information, please consult the AEQ Sales Department or authorized distributors.

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

ANNEXE B: Connecting PHOENIX STRATOS to ISDN National-1.

The following paragraphs offer some practical notes for configuring the ISDN TERMINAL ADAPTER (TA) in the UNITED STATES OF AMERICA and OTHER COUNTRIES using the NATIONAL-1 ISDN PROTOCOL:

The ISDN (Integrated Services Digital Network) is a telephonic connection system that enables its users to establish completely digital end-to-end communication. According to the number of 64-Kbps channels, there are two types of ISDN links: Basic Rate links (BRI) Primary Rate links (PRI):

- The **Basic links** enable two data connections or channels to be established simultaneously, each with a capacity of 64Kbps.
- The **Primary links** make it possible to establish 23 64-kbps channels in the United States of America and 31 64-Kbps channels in Europe.

To access ISDN services, the would-be user must subscribe to an ISDN telephone line to which a device can be connected, such as an audio encoder which is equipped with an ISDN Terminal Adapter that will perform all the connection establishment and maintenance management through ISDN digital telephone lines.

The PHOENIX STRATOS has an ISDN Terminal Adapter for Basic links (BRI), which gives it a communication capacity of two independent 64-kbps data channels.

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

- The **U interface** links the telephone exchange to the subscriber; it is a two-wire interface of the type used in analog telecommunications, and permits only a point-to-point connection (a single device connected to the interface).
- The **S/T interface** is obtained through the U interface by passing through a Network Terminator (TR1) installed on the subscriber's premises; it is a four-wire interface that permits the multipoint connection of up to seven devices.

In USA, telephone companies do not supply the TR1; it is up to users to install these devices if they want them. The most usual situation is for the TR1 not to exist, and for the access to be accomplished through the U interface.

In short, in USA the normal solution is access through the U interface unless the subscriber has requested or the telephone company has installed a TR1 of its own volition. (The TR1 is normally a little box secured to the wall with a power supply and a luminous indicator of some type, into which the two-wire outside line enters, offering RJ-45 outputs).

The universal TA (Terminal Adapter) included in the PHOENIX permits both types of connection. For that purpose, it is equipped with two ISDN connectors, one marked ETSI (RJ-45, 8 pins) for connection through an S/T interface, and the other marked ANSI (RJ-11, 6 pins) for connection through a U type interface. The type of connection used must be configured during the equipment set-up procedure.

Although the ISDN is intended to be a uniform standard, differences can be found among the protocols used in different countries. In USA, despite the presence of several protocols that are incompatible with each other, there is, fortunately, one protocol that is relatively standardized (called National-1), and which is usually available in all the telephone exchanges.

The Universal Terminal Adapter integrated into the PHOENIX STRATOS can operate with the NATIONAL-1 protocol (the most frequently used protocol in the USA). The choice of the desired protocol, in this case NATIONAL-1, must be configured during the equipment set-up.

In Europe and other geographic areas there is a protocol called EURO ISDN. There are minor differences among the EURO ISDN standards applied in different countries. These differences do not affect the PHOENIX STRATOS.

B1. Line Identification Numbers (DN's or Directory Numbers).

These are the telephone directory numbers assigned to the line.

In the **National-1** protocol there will be two numbers, each of which will be associated with one of the line channels. These numbers have to be programmed in the PHOENIX STRATOS unit so that it will work properly ("NUMBER LINE X" option), and thus each channel will only respond to the number associated with it (**NOTE:** In the United States the identification number that must be programmed in the terminal cannot contain the area code; it must contain only the seven digits of the local number).

If the identification numbers are incorrectly programmed in the codec, the unit may make outgoing calls but will not respond to incoming calls.

B2. SPID's (Service Profile Identification Numbers).

In the case of the NATIONAL-1 protocol, in addition to the line identification numbers (DNs), you will need to program SPIDs (Service Profile Identification Numbers) in the terminal. These are numbers that the service provider assigns in association with each of the two line channels. If the SPIDs are not configured correctly, the terminal will not work.

B3. Line characteristics for the USA. Line request model.

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

To ensure that the line characteristics are properly suited to their use with the PHOENIX STRATOS codec, **we recommend** using the following form for requesting the line:

LINE REQUEST FORM:

We request an ISDN Basic Rate Interface (BRI) line for an Audio Codec device.
We will provide the NT1 and need a U 2B1Q interface on a standard RJ-11 jack.
We support only the *National ISDN-1 Protocol*.
We need clear 64Kbps access. (56Kbps is not acceptable)
Long distance prefix (10XXX) must be enable.

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

B4. Connecting the PHOENIX STRATOS to the ISDN line.

If the interface to be used is of the U type (the most common type in the US), connect the line to the 6-pin RJ-11 base connector marked ISDN-ANSI in the back panel, using a 6-pin modular telephone cable (RJ-11 to RJ-11). Note that the point of connection to the U interface delivered by the telephone company may be a 6-pin RJ-11 type base or an 8-pin RJ-45 base (the standard accepts both types). If the connector provided is of the RJ-45 type, you can insert the 6-pin RJ-11 connector terminating the cable in the center of the 8-pin RJ-45 base.

ANNEXE C: Protocols associated with communications 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 clear-down.

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

N/ACIP

- Signaling, understood as connection initiating and finalizing 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 appendix is not intended to be a reference document for all the relevant technical matters, it should at least serve to give its readers an initial contact with these subjects that will ease the assimilation of the new working method over IP networks for the PHOENIX STRATOS user and, as a result, the use 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 versus packet switching.

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

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.

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

C1.1.1. Advantages:

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

- Once the circuit has been established, the parties can communicate with each other at the highest speed that the medium allows, without having to share 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. Advantages:

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

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.

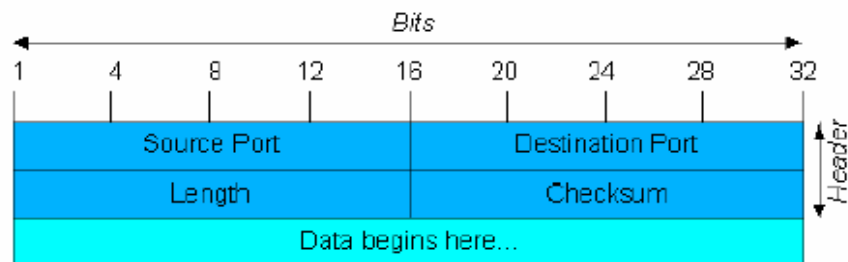
C2. IP Protocol.

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, we recommend consulting:

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

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

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.

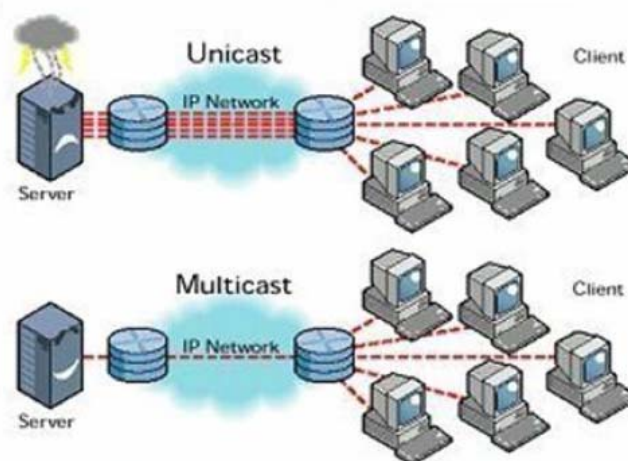
Another possible IP address classification can be made according to address validity:

- **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.

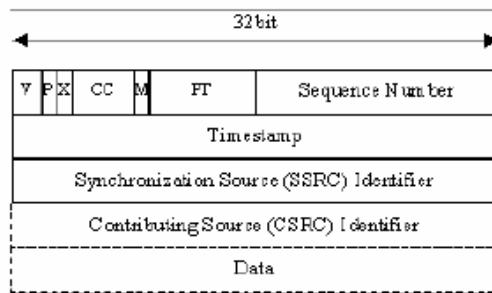
C3. RTP Protocol.

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

The IP ports defined for its use are 5004 (RTP) and 5005 (RTCP).

The functions of the RTP/RTCP protocol are:

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



RTP Header

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

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

C3.1. PHOENIX STRATOS default configuration.

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

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

These values can be modified from its internal menu.

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 finalizing interactive user sessions involving multimedia elements such as video, voice and instant messaging.

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

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

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

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

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

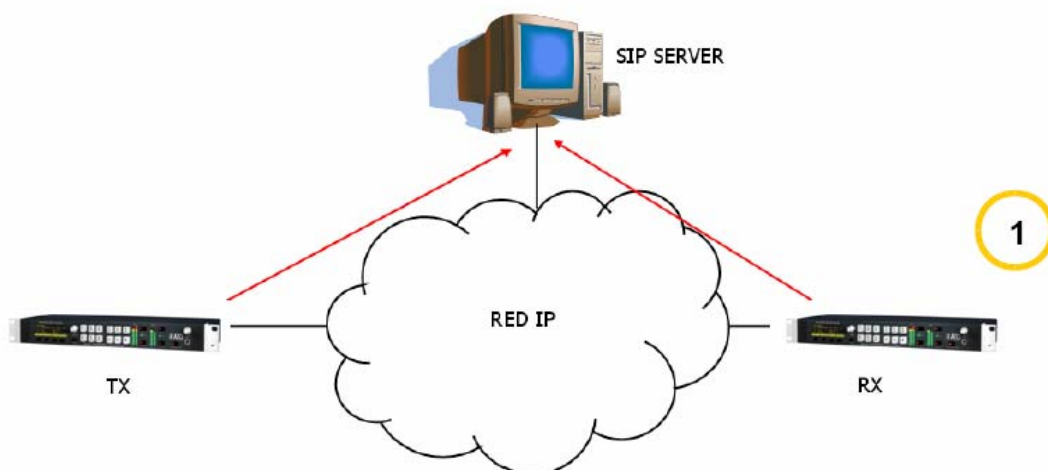
SIP also contributes register functions that enable the user to inform the proxy servers of his or her location.

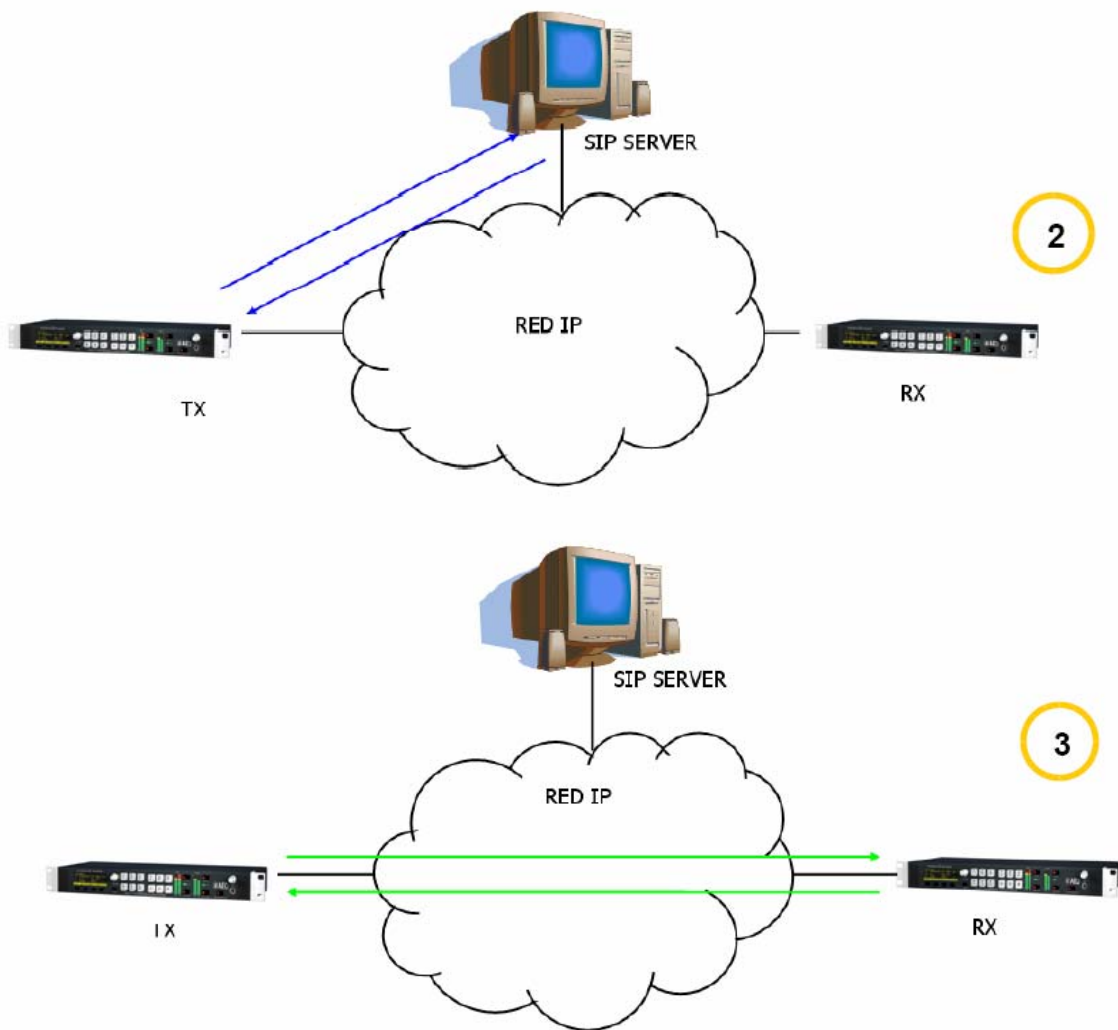
For complete information on the SIP protocol, we recommend consulting:
<http://tools.ietf.org/html/rfc3261>

C4.1. Working modes.

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

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





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

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

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

C4.2. Possible work scenarios.

Depending on the type of network to which the PHOENIX STRATOS 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 STRATOS operates perfectly with an associated SIP server and equipment identifiers of the name@domain type (PROXY SIP), and even without an associated SIP server with a URI of the name@IP-address type (DIRECT SIP) if the device on the opposite end of the communication circuit also has an IP address of the same type.

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

- Dynamic public IP addresses, corresponding to use an Internet access by means of a single workstation router and a dynamic IP (the most usual). Allows the use of URIs of the name@domain (PROXY SIP) or name@IP-address (DIRECT SIP) type, but it is advisable always to work with an associated SIP server (PROXY SIP), since the IP address assigned to the equipment may change each time the user powers up the unit.
- Private IP addresses, both static and dynamic, corresponding to connections in a local network with several workstations, that access to the Internet through a router with NAT. Those do not allow the use of URIs of the name@IP-address type because the IP address of the identifier is not public, and is only valid in the section of the network to which it has been assigned; it lacks a universal meaning. In this case the use of an associated SIP server and a STUN server is imperative to get past the NAT (Network Address Translation) implemented in the router that acts as an interface between the private network and the public one. See section NAT TRAVERSAL (3.1.5.2.4).

C4.3. PHOENIX STRATOS default configuration.

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

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

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

PHOENIX STRATOS is supplied preconfigured with 2 users registered in both servers:

- PROXY SIP “AEQ” and “AEQ 2” Authentication Users: phxstratos_XXX y phxstratos_XXY (where $Y=X+1$)
- PROXY SIP “AEQ” and “AEQ 2” Authentication Passwords: 8 digits alphanumeric passwords

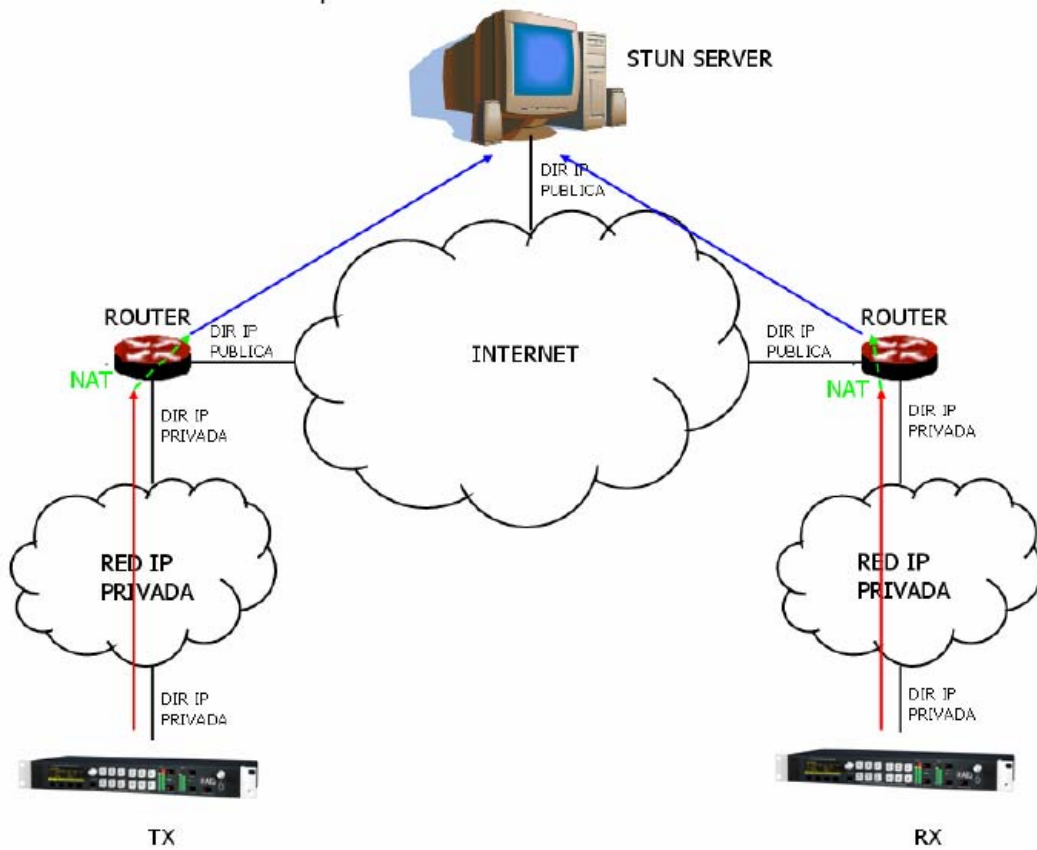
This configuration is the right one for working with any of both AEQ's SIP servers.

C5. STUN protocol.

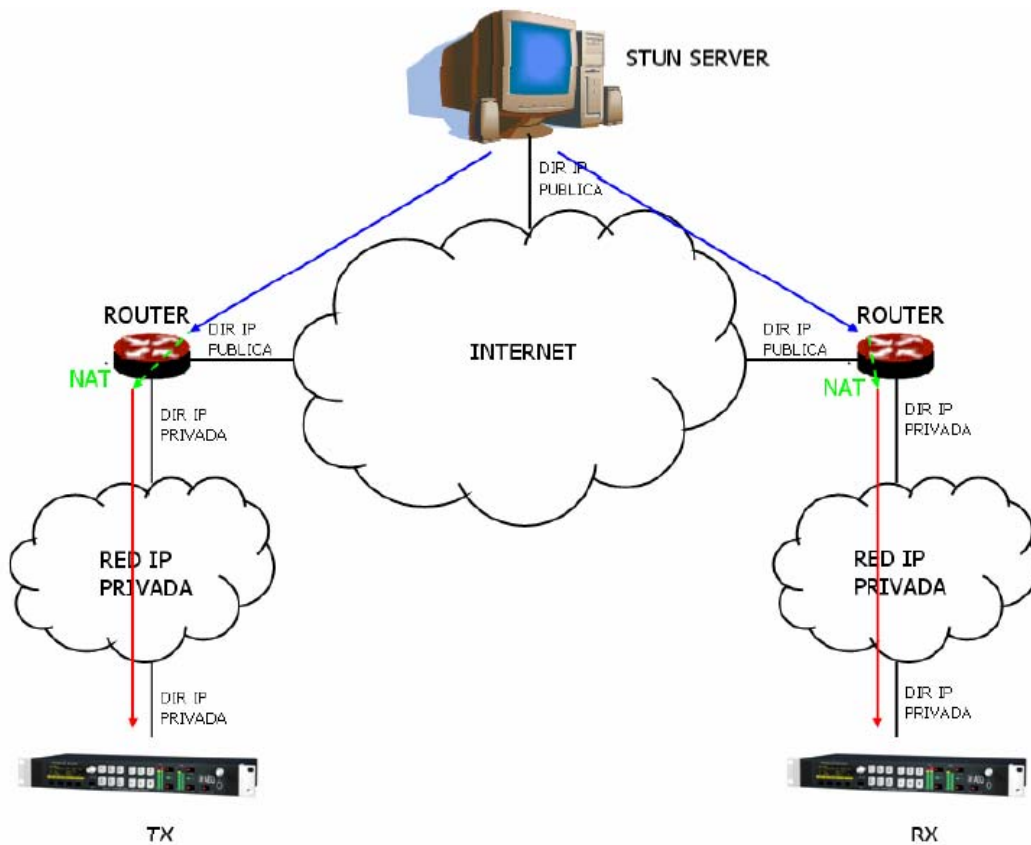
STUN (Simple Transversal of UDP over NATs) is a network protocol of the client /server type that allows NAT clients to find their public IP address, the type of NAT where it is located and the Internet port associated with the local port through NAT. This information is used to configure a UDP communication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

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

PHOENIX STRATOS 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 STRATOS to identify its position within the SIP server. This protocol is used in “AUTO3” and “AUTO4” NAT TRAVERSAL modes (see section 3.1.5.2.4).



Registration in a STUN Server



Notification from the server of the corresponding NATs detected

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

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

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

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

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

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

ANNEXE D: Ports used by PHOENIX equipment.

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

1 - Output permissions in router+firewall: Phoenix unit will send packets to different servers and/or other units (each one will use a different port). Therefore, firewall will have to allow that packets from Phoenix unit IP are sent towards the following ports:

UDP 53	DNS protocol (servers names resolution)
UDP 162	SNMP Traps protocol (alarms)
UDP from 5004 to 5010 (it may vary)	RTP protocol (audio packets towards the remote unit, ports depend on the remote unit network, not Phoenix one)
UDP 3478 and 3479	STUN protocol (server for getting the public IP of the private network)
UDP from 5060 to 5062 (it may vary)	SIP protocol (SIP signalling packets, port depends on the remote unit network, not Phoenix one)
UDP 50000 to 60000	VUmeters protocol (only when remote control PC is not in Phoenix network)
TCP 4422 (it may vary)	remote control protocol (only when Phoenix must be connected to PC and PC is not in its network. Port depends on PC, not on Phoenix)

2 - Input permissions in router+firewall: Phoenix unit will have to be able to receive packets from units installed out of the private network. Therefore, firewall will have to allow that packets sent to Phoenix unit IP to the following ports are received:

UDP 161	SNMP protocol (monitoring)
UDP from 5004 to 5010 (it may vary)	RTP protocol (audio packet toward the remote unit, ports depend on Phoenix configuration)
UDP from 5060 to 5062	SIP protocol (SIP signalling packets)
TCP 80	HTTP protocol (only when PC must be connected to Phoenix web server for a firmware upgrading, for instance)
TCP 4422	remote control protocol (only when PC must be connected to Phoenix and PC is not in its network)

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

ANNEXE E: Application notes guide.

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

All notes are available in electronic format in the CD furnished with the unit.

E1. Internet connection using standard cable access.

Application note NA0A.

Phoenix Studio audiocodec directly connected to Internet by means of a dedicated cable-Modem with DHCP. SIP call using AEQ SIP Proxy.

Application note NA0B.

Same case as NA0A, but adapted to Phoenix Mobile.

Application note NA0C.

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

Application note NA0D.

Same case as NA0C, but adapted to Phoenix Mobile.

Application note NA0E.

Same case as NA0C, but calling in SIP DIRECT but with no SIP Proxy involved.

Application note NA0F.

Same case as NA0E, but adapted to Phoenix Mobile.

E2. Special applications using different kinds of Internet physical accesses, or point to point connections.

Application note NA1.

Connecting a Phoenix (Studio and Mobile) to Internet through a PC via a WiFi network.

Application note NA2.

Connecting two Phoenix Mobile units using a BGAN satellite link.

Application note NA3.

Connecting two audiocodecs (Phx. Studio – Phx. Studio & Phx.Mobile – Phx. Studio) using a private WiMAX network.

Application note NA4.

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

Application note NA5.

Connecting a Phoenix Mobile to Internet using a 3G router.

Application note AN5B.

Connecting Phoenix IP audiocodecs to 3G/4G networks.

ANNEXE F: Additional information.

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