



Multiboard Communications System

MPEG Layer III

USER'S MANUAL

ED 03/07

V 1.2 - 13/04/2007

Firmware Versions: Controller 2.20
ISDN - V35/X21 board (Micro 3.03 / DSP 2.05 / TA 6.00)
Hybrid board (DSP 1.00)
TCP/IP board (Rabbit 1.03 / DSP 2.05) or higher.

Software Versions: [E@sy](#) Server 1.70.003 / RTC 1.70.006 / FU 1.70.001 or higher.

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1. EQUIPMENT DESCRIPTION.

1.1. Basic design concepts.

Course is a modular system with a capacity of up to 10 communication boards in a 4-units rack with redundant power supply. There are three kinds of boards: ISDN or V35/X21 communication boards, TCP/IP communication boards and digital hybrids (analog line).

The AES-EBU inputs and outputs are optional. It also offers an ISDN universal terminal adapter for ISDN communication boards to easily connect to S and U interfaces.

The system configuration is:

- Up to 10 DSP front cards (processor board). This is the card that processes the audio. The “engine” of each module, it can include 1 or 2 AES/EBU interfaces.
- Up to 10 rear cards that could be:
 - ISDN - V35/X21 communication board. Made up of a two (stereo) channels audio interface card, with analog inputs and outputs, DB9 connectors to incorporate one or two AES/EBU inputs/outputs and 24-bit AD/DA converters. It incorporates input and output transformers, independent sample rate converters (SRC) at each input and output, with 1:3 and 3:1 range and 24 bits. It also has a V.35/X.21 interface, which allows connections at 64, 128 or 256 Kbps. Also incorporates a port for the auxiliary data channel. It is possible to add one S interface & U interface terminal adapter per rear card, valid for EURO ISDN (ETSI) and American NATIONAL-1 (ANSI).
 - Telephone hybrid board for connection through conventional telephone lines (POTS). It provides pulses and/or DMTF dialing function, as well as incoming signal detection (ring) and incoming call identification functions. Hybrid boards can manage simultaneously and independently two analog telephone lines.
 - TCP/IP communication board. Acts as a communication interface for encoded audio (one or two channels and several encoding formats) through Ethernet with TCP and/or UDP protocol. The Ethernet connection is at 10/100 Base T through a RJ45 connector. In case of using two channels, the maximum baud rate per channel is 64Kbps. In case of one channel, the maximum baud rate is 128Kbps.
- 19” rack, 4U with capacity for 10 cards and their respective complementary cards. It includes the system control card, the RS-422 and RS-232 remote control interfaces, the AES/EBU external sync input (AES11) and its through output, and two 90-250 V, 50-60 Hz autorange power supplies.

1.2. Functional specifications and special features.

- Each ISDN - V35/X21 communication board is a multi-standard dual two-way digital audio codec for ISDN lines, with Intercommunication. It can make two independent calls at 64 kbps or one at 128 kbps, using U or S interfaces; while with the V.35 interface, it can make one connection at 64 kbps, one at 128 kbps or one at 256 kbps. In the 64 kbps modes, it is possible to have separate encoding algorithms for the B channels.
- Each telephone hybrid board for connection through conventional telephone lines (POTS) has two communication lines that can be operated simultaneously and independently.
- Each TCP/IP communication board is a multi-standard dual two-way digital audio codec for connection through Ethernet with TCP and/or UDP protocol. It can make one Ethernet connection with one or two independent audio channels.
- Connection to an RS-422 multi-station network, for simultaneous control and upgrading of up to 100 units.
- Off-hook (or audio connection for TCP/IP boards) automatic or manual mode selection.
- Alphanumeric phone and IP addresses book.
- Configuration of audio encoding and editing modes can be done simply and intuitively, using the control software in real time.
- Configurable audio inputs and outputs: analog at line level and digital in AES/EBU format, with 24-bit and SRCs (Sampling Rate Converter) at all the digital inputs and outputs.
- Generation of DTMF tones, for control of remote equipment.
- Flash EPROM-type internal memory, to upgrade your equipment with future versions with no need to open up the unit.
- Members of the **E@sy** family: from a multiple workstation network, they can control a set of COURSE codecs and other **E@sy** equipment units, making a high performance system with powerful functions.
- High communication capacity in a small space: in four vertical units, up to 20 mono or 10 stereo circuits.
- For ISDN communication boards, built-in ISDN terminal adapter (optional) with a dual port interface 'S' and 'U' which is compatible with ETSI (Euro ISDN) and ANSI (NATIONAL-1), with the possibility of communication at 64 y 128 Kbps. It is part of the World Wide Ready Series family, which includes interfaces for connections to the different world communications networks.
- Regarding TCP/IP communication boards, the IP connection can be established within a private network such as a LAN, a public network, that unites two or more LANs or by means of the Internet. Once the IP connection is established, the audio stream is totally bidirectional between the COURSE equipment. The system is totally independent from the hardware of the network, as long as an IP connection between the equipment is provided. It is possible to use wireless routers, wired networks, satellite networks, etc. It is important to have fixed IP's for the equipment, and to avoid dynamic IP's that are automatically assigned by a DHCP server.
- It is an investment suited to real needs, since it is possible to configure the use of V35/X21 and ANSI/ETSI interfaces, a number of channels per board (one or two), a number of audio interfaces and AES-EBU boards.

- Double channel in all the 64 Kbps coding modes (except for MPEG Layer III).
- For telephone hybrid boards, echo cancellation capability and frequency extender available to improve the audio quality of telephone analog line calls.

1.3. Encoding algorithms.

Main function of an AUDIOCODEC is to establish one or two bidirectional audio channels between distant points for receiving a telephone chronicle, for transmitting with a high quality a sports commentary voice or for establishing an audio link between two studios. In the remote site, a telephone or another AUDIOCODEC will receive the encodec audio and will decode it into audio signal.

The requirements of the communication channel in each of these applications are different and depends on the type of audio signal to be transmitted (voice or music, mono or stereo), required audio bandwidth, delay time, etc.

All available encoding algorithms features are compared in the following figure. The **blue** ones are available only for ISDN - V35/X21 communication boards.

Encoding Algorithms	Transmission rate	Audio format	Audio bandwidth	Approximate delay time*	Available in channels	Ancillary datachannel
VOICE ALGORITHMS – For transmitting voice with telephone quality.						
G.711 LAW-A (EUROPE)	64Kbps	Mono	3.5kHz	37 mseg	L1 & L2	NO
G.711 LAW-μ (USA & JAPAN)	64Kbps	Mono	3.5kHz	37 mseg	L1 & L2	NO
SPECIAL ALGORITHM AUDIO+DATA						
AEQ-3.5 32kb	64 (32+32)	Mono	3.5kHz	25 mseg	L1 & L2	YES
STANDARD ALGORITHMS – For transmitting voice with medium quality.						
G.722 STAT	64Kbps	Mono	7kHz	21 mseg	L1 & L2	NO
G.722 H.221	64Kbps	Mono	7kHz	20 mseg	L1 & L2	YES
MPEG 64 ALGORITHMS – For transmitting voice in high quality and music in medium quality.						
MPEG L2 64 24 kHz	64Kbps	Mono	11kHz	450 mseg	L1 & L2	YES
MPEG L2 64 32 kHz	64Kbps	Mono	10.5kHz	340 mseg	L1 & L2	YES
MPEG L2 64 48 kHz	64Kbps	Mono	10kHz	240 mseg	L1 & L2	YES
MPEG L3 64 48 kHz	64Kbps	Mono	15kHz	235 mseg	Only L1	NO
SPLIT 64 ALGORITHMS – To transmit and receive using different algorithms.						
MP3 SPLIT 64 cMP3 dG722	64Kbps	Mono	Tx 15kHz Rx 7kHz	Tx 235mseg Rx 21mseg	Only L1	NO
MP3 SPLIT 64 cG722 dMP3	64Kbps	Mono	Tx 7kHz Rx 15kHz	Tx 21mseg Rx 235mseg	Only L1	NO
128 MONO ALGORITHMS – For transmitting voice and music (mono) in very high quality.						
AEQ LD-2	128Kbps	Mono	15kHz	22 mseg	L1 + L2	NO
MPEG L2 128 MONO 48 kHz (J.52)	128Kbps	Mono	20kHz	320 mseg	L1 + L2	YES
MPEG L2 128 MONO 48 kHz (IMUX)	128Kbps	Mono	20kHz	240 mseg	L1 + L2	YES
MPEG L3 128 MONO 48kHz	128Kbps	Mono	15kHz	550 mseg	L1 + L2	NO
128 DUAL ALGORITHMS – For transmitting two voice channels in very high quality.						
MPEG L2 128 DUAL 32 kHz (J.52)	128Kbps	Dual	10.5kHz	470 mseg	L1 + L2	YES
MPEG L2 128 DUAL 32 kHz (IMUX)	128Kbps	Dual	10.5kHz	345 mseg	L1 + L2	YES
128 MPEG STEREO ALGORITHMS – For transmitting music in medium and high quality.						
MPEG L2 128 JOINT 48kHz (J.52)	128Kbps	Stereo	15kHz	300 mseg	L1 + L2	YES
MPEG L2 128 JOINT 48kHz (IMUX)	128Kbps	Stereo	15kHz	240 mseg	L1 + L2	YES
MPEG L3 128 STEREO 48kHz	128Kbps	Stereo	15kHz	550 mseg	L1 + L2	NO
256 MPEG STEREO ALGORITHMS – For transmitting music in very high quality (Only through V35).						
MPEG L2 256 STEREO 48kHz	256Kbps Only V.35	Stereo	20kHz	110 mseg**	-	YES

* “Delay Time” measures the time that goes between audio is feed into one AUDIOCODEC input until it reaches remote AUDIOCODEC output. It implies encoding time plus transmission time plus decoding time. This measures had been obtained using an ISDN switchboard that has very low transmission delay. So real time in a long distance communication will be higher than this. Ancilliary Data channel usage does not affect delay time.

** This value includes encoding+decoding time. Line transmission time must be added.

In the following page, the different encoding algorithms are explained.

In the following Coding Algorithms, the COURSE both, transmits and receives on the same Encoding Algorithms:

- **G.711:** By using this algorithm it is possible to generate and receive voice calls, and to establish communication through ISDN with telephone numbers or lines pertaining to the basic and analogue telephone network (PSTN).
- **G.722:** This mode is widely accepted for bi-directional communication, thanks to the low delay induced, it is frequently used for voice-overs and commentator applications such as sport coverage.
 - The **G.722 H221/H242** mode provides a bi-directional ancillary data channel simultaneously with the Audio transmission.
 - The **statistical G.722** mode is performing a real-time Audio processing of the transmitted Audio signal that allows a continuous bandwidth adjustment, which improves the quality of some Audio signals, for example the voice of a sport commentator. In this case no ancillary data channel is needed.
- **MPEG Layer II:** Very well known and accepted encoding mode, used when delay is not a factor, as MPEG modes always have higher delay than G.722 modes. The COURSE includes MPEG LII for mono, dual or stereo transmission.

The equipment implements two different framing modes for the transmission of 128 KBPS programs in MPEG Layer II, combining the two 64 KBPS B-channels of the ISDN line:

 - **J.52 Multiplexing:** J.52 is the international standard that defines how to frame and transmit a Multi-channel. In the COURSE, two 64 KBPS channels are framed and transmitted.
 - **IMUX (Recommended mode):** Framing for 128 KBPS modes, implemented for compatibility with other equipment that are available on the market. This framing mode also offers a lower delay than the J.52 framing mode.
- **MPEG Layer III:** Very well-known and accepted algorithm that presents better Audio bandwidth at lower bit rates but at the cost of a longer encoding and decoding delay. The COURSE includes MPEG LIII for transmission in mono or stereo.
 - In order for MPEG Layer III at 128kbps to function properly the IMUX framing mode must be selected.
 - In the transmission modes of MPEG Layer III at 64kbps only channel 1 can be used.
 - In the MPEG Layer III modes it is not possible to generate DTMF tones.
 - In the MPEG Layer III it is possible to use different modes simultaneously within a single transmission, for example MONO and STEREO within the same communication.

MPEG Layer-3 audio coding technology licensed from Fraunhofer IIS and Thomson

- **AEQ LD-2:** AEQ proprietary mode (based on the previous AEQ LD-EXTEND) combining the low delay of the G.722 with the greater bandwidth of MPEG, thus optimizing both modes. It can only be used if there is equipment with the same encoding mode at the other end, i.e., another COURSE, a COURSE, the AEQ model ACD-5001 (stationary) or an MPAC-02.
 - If you have older AEQ CODECS that implement the **AEQ LD-EXTEND** and want to make them compatible with the COURSE in this algorithm, visit our web-site <http://www.aeq.es/> and upgrade them to the latest firmware version available.

In the SPLIT modes, the transmission (encoding) is independent of the reception (decoding). This system allows you to receive a high quality Program Audio Signal from an event with a relatively low quality Return Audio Signal with low delay.

- **SPLIT MODE:** Allows to transmit in MPEG L3 64 MONO 48kHz (high quality) and to receive in G722 STAT (low delay) using only a single B-channel of the line ISDN line. The second B-channel of the ISDN line cannot be used. Evidently, the Audio Codec that receives the signal will have to carry out the inverse operation, to receive in MPEG L3 64 MONO 48kHz and to transmit in G722. To set up an COURSE for this mode, "cMP3 dG722" should be selected and the remote COURSE should be set to "cG722 dMP3".

In the Audio Codec boards, the encoding modes can be changed once a communication has been established and without the need of interrupting it thanks to the Terminal Adapter implemented in the equipment. When a connection at 128kbps is changed for one of 64kbps, channel 2 will be disconnected.

1.4. Available modules.

- **COURSE-42:** 19" rack, 4U with capacity for 10 front cards, 10 rear cards and their respective complementary cards. Includes the system control card, RS422 and RS232 remote control interfaces, AES/EBU (AES-11) sync port and its through output. Also includes two autorange power supplies (90-250V, 50-60Hz) with automatic power factor correction.
- **COURSE-DSP:** Audio processor card, this is the engine and indispensable element on each module. Ready to include one or two AES/EBU Digital Audio I/O boards (optional). COURSE-DSP is the basic front board.
- **COURSE-ISDN:** ISDN communication board made up of 2 modules:
 - **COURSE-2C:** Two-channels (stereo) audio interface card. Each incorporates two each balanced analog inputs and outputs per transformer and two DB9 connectors to incorporate one or two AES/EBU optional modules. Equipped with 24-bit AD/DA converter.
 - **COURSE-V35:** Communications interface card with V.35/X.21 interface. Allows connection at 64Kbps, 128 Kbps or 256kbps. This card also incorporates a port for the auxiliary data channel at binary rates configurable in the encoding modes that support this capability.
- **COUSE-HYBRID:** Telephone hybrid board for connection through conventional telephone lines (POTS). It provides pulses and/or DMTF dialing function, as well as incoming signal detection (ring) and incoming call identification functions. Hybrid boards can manage simultaneously and independently two analog telephone lines.
- **COURSE-IP:** TCP/IP communication board. Acts as a communication interface for encoded audio (one or two channels and several encoding formats) through Ethernet with TCP and/or UDP protocol. The Ethernet connection is at 10/100 Base T through a RJ45 connector. In case of using two channels, the maximum baud rate per channel is 64Kbps. In case of one channel, the maximum baud rate is 128Kbps.

Optional modules:

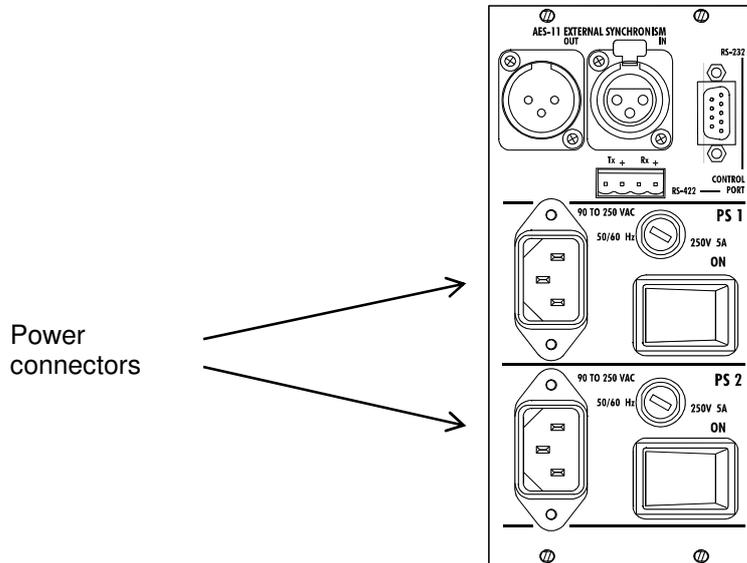
- **COURSE-AES/EBU:** AES/EBU (AES-3) input/output card. Incorporates transformers, independent sample rate converters (SRC) at each input and output, with 1:3 and 3:1 range and 24 real bits with no truncation. It supports binary rates of 16 to 48 KHz. SRC dynamic range is 128 dB and THD+noise is at 1 KHz of -117dB. If two COURSE-AES/EBU cards are installed on one module, the latter includes two independent mono inputs and outputs, allowing different sampling rates. If only one is installed, a stereo or mono input and output can be incorporated.
- **COURSE-TA:** Communications interface card with 'S' and 'U' universal terminal adapter and V.35/X.21 interface. Adds to COURSE-V35 board capabilities, a universal terminal adapter, which is valid for both EURO ISDN (ETSI) and American NATIONAL-1 ISDN (ANSI).

See physical location of the different modules in chapter 8.8.

2. EQUIPMENT POWER SUPPLY.

2.1. Equipment power supply.

As it has two redundant power supplies, a network cable must be connected to each one. These power connectors are located on the unit's back panel.



The equipment receives power through the connectors designed for that purpose, using the cables supplied. The equipment may be operated with alternating voltage of between 90V and 250V at 50 or 60 Hz. Maximum consumption is approximately 120 VA.

2.2. Turning on the equipment.

Before connecting it, make sure the power switches, located on the back of the unit, are in the **OFF** position.

Once the power cables are properly connected to the equipment, you can connect the power cables to the electrical power network.

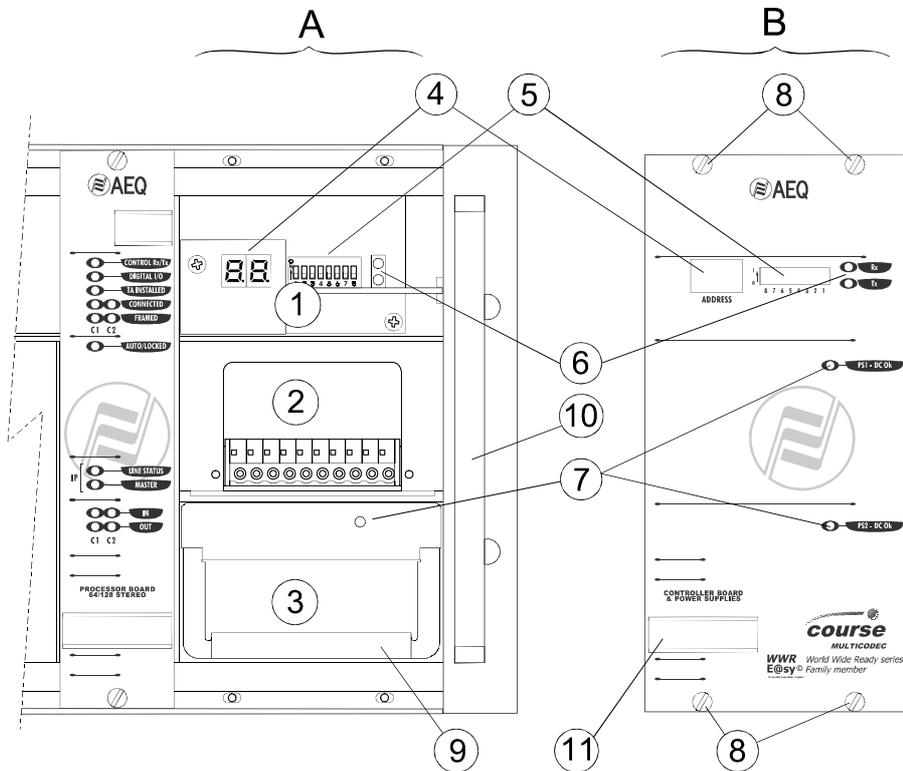
Activate the power switches (**ON** position). If all instructions have been correctly followed, each power supply's POWER ON LED will light up, indicating that the equipment is receiving power.

3. INSTALLATION AND CABLING.

3.1. Power Supply and Control Module.

To clarify the cabling and installation process, it is necessary to become familiar with the connectors and configurable elements on front and back panel.

3.1.1. Front panel description.



A.- Control Module / Power Supplies.

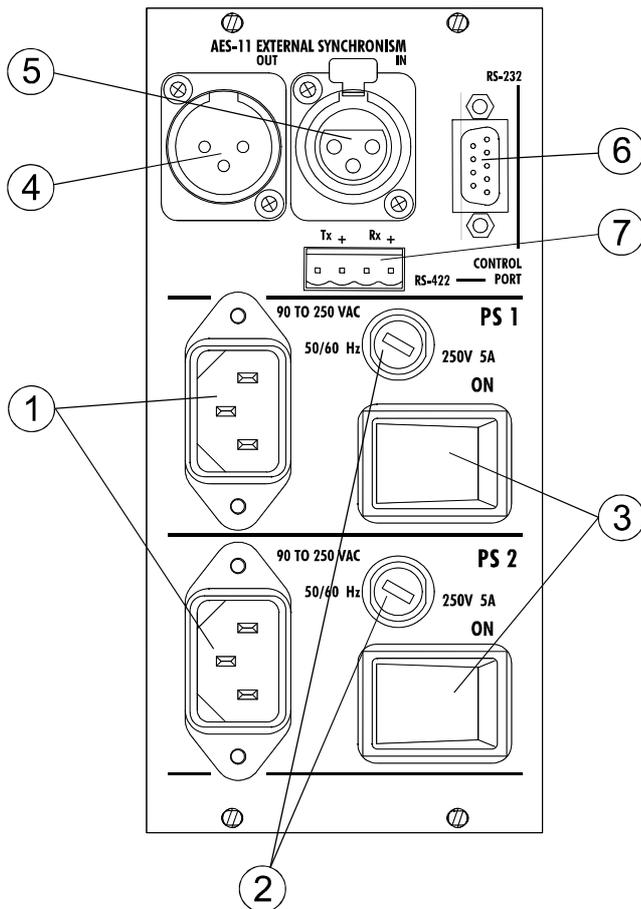
B.- Front Panel.

- 1.- Controller card.
- 2.- Power supply 1 (PS 1) housing.
- 3.- Power supply 2 (PS 2) inserted.
- 4.- Address indicator display .
- 5.- DIP address configuration microswitches.
- 6.- LEDs for communication with the PC.
- 7.- POWER ON LED indicator.
- 8.- Screws to secure module.
- 9.- Power supply pull handle.
- 10.- Rack right handle.
- 11.- Label Holder.

The DIP configuration microswitches are numbered from 1 to 8 and are accessible on the front panel. Each microswitch's function is to establish the address of the equipment, which is specified between 0 and 99. The Tx and Rx LED's indicate that data are being sent or received from the control PC.

The position of the microswitches should be established before turning on the equipment. Any modification to the microswitches once the equipment is started up will not be taken into account.

3.1.2. Back panel and connections description.



- 1.- Power cable connectors. Cables supplied with the equipment.
- 2.- Fuse holder.
- 3.- On/off switches.
- 4 and 5.- XLR AES/EBU input/output sync connectors.

This input can be used for linking internal AES/EBU clock (48kHz by default) to an external clock source (for example 32kHz, 44.1kHz or 96kHz). Installed AES/EBU cards will use this clock.

The connection for both the input (XLR female) and for the through output (XLR male) is:

<u>Contact</u>	<u>Signal</u>
1	GND
2	Sync V+
3	Sync V-

6.- RS-232 **E@sy** control port.

The RS-232 control port gives access to a RS-232/RS-422 internal converter, which makes it possible to have a direct connection with the serial port of the control PC using a standard RS-232 cable (DB-9 male to DB-9 female pin to pin). In this case, the RS-422 port allows you to connect several **E@sy** equipments into a network. Active contacts are:

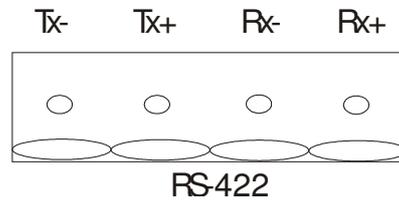
	COURSE		PC
TX	2	----	2
RX	3	----	3
GND	5	----	5
RTS	7	----	7

COURSE

11

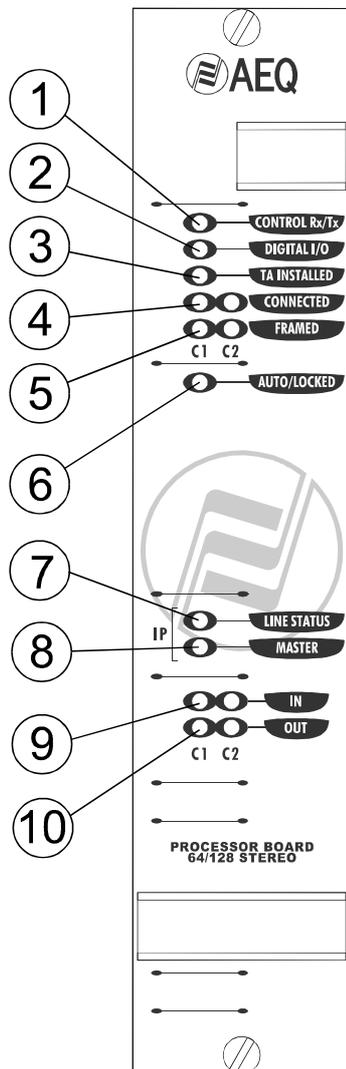
7.- RS-422 E@sy control port.

It has a four-contact connector, two for transmitting and two for receiving. On one hand, the RS-422 control port allows you to connect the equipment with a PC (you will also need a RS-232/RS-422 standard external converter to make this connection through the serial port of the control PC) and, on the other hand, allows you to connect several equipments in parallel in order to form an E@sy network. The connection is:



3.2. ISDN - V35/X21 communication board.

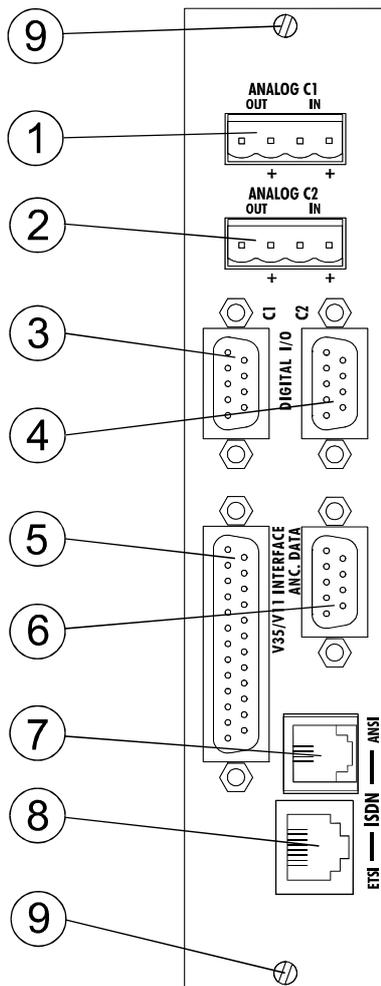
3.2.1. Front panel description.



- 1.- E@sy bus activity indicator.
- 2.- Digital I/O indicator.
- 3.- Internal Terminal Adapter installed indicator.
- 4.- Connected communication indicator, per channel.
- 5.- Sync indicator, per channel.
- 6.- Automatic coding synchronization indicator.
- 7.- TCP/IP connection establishment indicator (only for TCP/IP modules).
- 8.- Master module indicator (only for TCP/IP modules).
- 9 and 10.- Input/output audio presence indicator, per channel.

Front module is the same for **ISDN - V35/X21** and **TCP/IP** communication boards (although firmware versions are different).

3.2.2. Back panel and connections description.



1 and 2.- ANALOG C1 and ANALOG C2 input and output connectors.

These connectors are “Hartman” type 4 Pin Connectors, two for Input and two for Output.

3 and 4.- DIGITAL I/O C1 and C2 digital input and output connectors.

These connectors are 9-Pin DB9 Female for the AES/EBU Inputs and Outputs. The Pin assignment for these connectors is:

<u>Pin no.</u>		<u>Connections</u>
1	-----	Rx N (V- input)
6	-----	Rx P (V+ input)
2	-----	Tx N (V- output)
7	-----	Tx P (V+ output)
5	-----	GND (ground)

- *When using the 64Kbps modes, the AES/EBU signal from channel 1 will be in connector C1(mono signal) and AES/EBU signal from channel 2 will be in connector C2 (mono signal).*
- *When using the 128Kbps or 256Kbps (mono, stereo or dual), the AES/EBU signal will only be in C1(mono or stereo signal).*

5.- V35 interface. DB25 connector.

It accepts two types of external TA:

- V35 (V11)/X21 interface: Uses unbalanced asynchronous control signals, while the synchronous clock and data signals are balanced.
- Physical level interface, compatible with X.21 / RS422 protocol levels.

The pin-out diagram and the signals available in the DB-25 provided in the AEQ COURSE are as follows:

Physical Level	Signal	Pin
RS 422 Bal	Clock (RCX) V+	13
RS 422 Bal	Clock (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

The COURSE can be configured from real time control software to choose if DCD signal is going to be used (DCD CONTROLLED) or ignored (PERMANENT). In this case, it would be necessary to connect only Tx, Rx and Clock signals.

CAUTION:
Unplugging the v.35 interface when power is on could damage COURSE internal circuits.

To connect the COURSE to an external TA, it will be necessary checking the external TA user's manual to make sure its pin arrangement corresponds exactly, due to the variety of connectors that a TA may have.

The pinout showed below is an example and corresponds to ISO-2593 standard in a MRAC-34 connector:

CCITT	DESCRIPTION	DTE	DB25		Terminal Adapter MRAC-34 connector	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 need to be configured. In those cases, check this parameters:

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

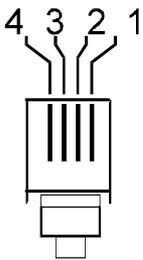
6.- Auxiliary data channel. RS232 port.

There is one DB-9 female chassis connector per module for auxiliary data transmission through the interface with an external computer and RS 232 protocol. A male-female cable should be used, and connected to your PC's port. The active contacts are:

	Course		PC
TX	2	----	2
RX	3	----	3
GND	5	----	5
RTS	8	----	8

Read more about datachannel at the end of this manual.

7.- RJ11 connector for ISDN line U interface (common in USA and other countries).



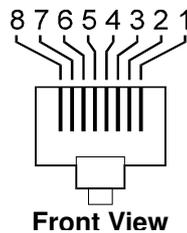
The RJ11 connector joins the terminal adapter included in the equipment to the ISDN network access point (network termination). This connector's pin arrangement is standard, and is shown as follows.

Use the cable supplied with the equipment to connect it to the ISDN jack.

Cable number		Connections
1		
2	-----	Lin V+
3	-----	Lin V-
4		

8.- RJ45 connector for ISDN line S interface (used by the Euro ISDN).

The RJ45 connector joins the terminal adapter included in the equipment to the ISDN network access point (network termination). This connector's pin arrangement is standard, and is shown as follows.



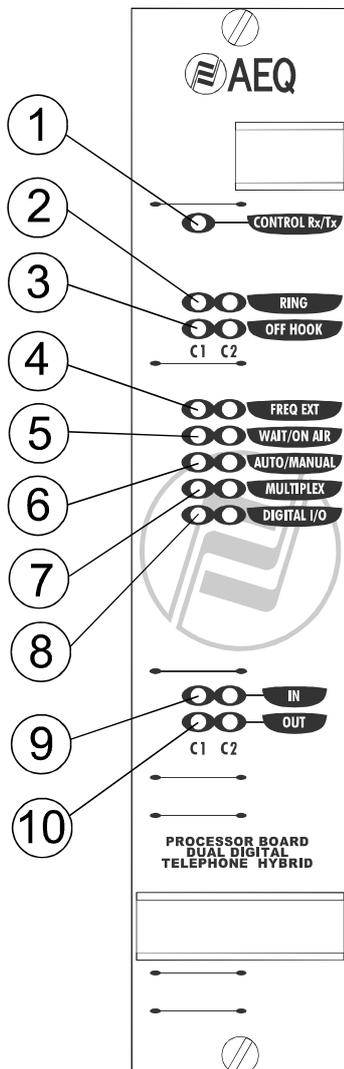
Use the cable supplied with the equipment to connect it to the ISDN jack.

Cable number		Connections
1		
2		
3	-----	Tx V+
4	-----	Rx V-
5	-----	Rx V+
6	-----	Tx V-
7		
8		

9.- Lock Screws for the Module

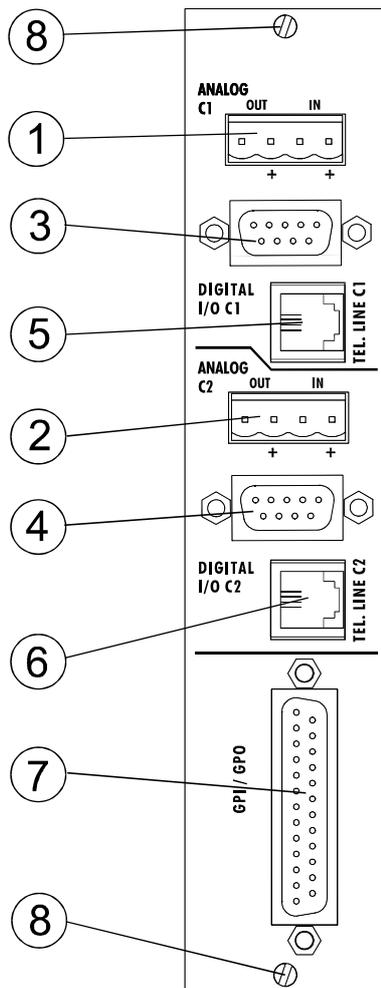
3.3. Hybrid board.

3.3.1. Front panel description.



- 1.- E@sy bus activity indicator.
- 2.- Incoming call indicator, per channel.
- 3.- Off-Hook indicator, per channel.
- 4.- Frequency Extender mode indicator, per channel.
- 5.- Wait/On Air mode indicator, per channel (lights on 'On Air' mode).
- 6.- Automatic/Manual Off-Hook indicator, per channel (lights on automatic mode).
- 7.- Active multiplex mode indicator.
- 8.- Digital I/O indicator.
- 9 y 10.- Input/output audio presence indicator, per channel.

3.3.2. Back panel and connections description.



1 and 2.- ANALOG C1 and ANALOG C2 input and output connectors.

These connectors are “Hartman” type 4 Pin Connectors, two for Input and two for Output.

3 and 4.- DIGITAL I/O C1 and C2 digital input and output connectors.

These connectors are 9-Pin DB9 Female for the AES/EBU Inputs and Outputs. The Pin assignment for these connectors is:

<u>Pin no.</u>		<u>Connections</u>
1	-----	Rx N (V- input)
6	-----	Rx P (V+ input)
2	-----	Tx N (V- output)
7	-----	Tx P (V+ output)
5	-----	GND (ground)

5 and 6.- RJ11 connector for analog telephone line.

4 3 2 1 This connector's pin arrangement is standard, and is shown as follows:



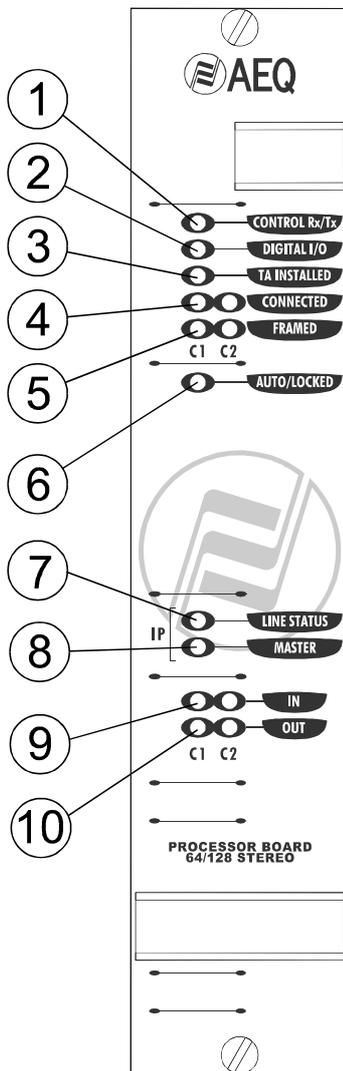
Cable number	Connections
1	
2	----- Lin V+
	3
	----- Lin V-
4	

7.- GPI/GPO connector.

8.- Lock Screws for the Module.

3.4. TCP/IP communication board.

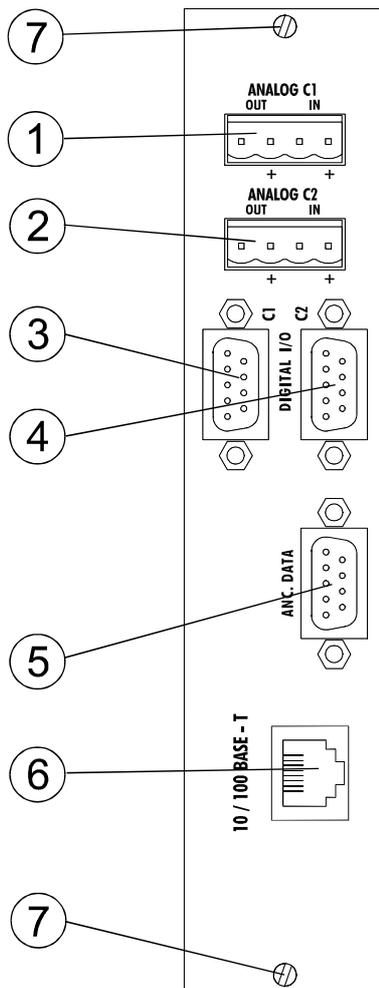
3.4.1. Front panel description.



- 1.- E@sy bus activity indicator.
- 2.- Digital I/O indicator.
- 3.- Internal Terminal Adapter installed indicator (only for ISDN - V35/X21 modules).
- 4.- Established audio connection indicator, by channel.
- 5.- IP connectivity indicator.
- 6.- Automatic coding synchronization indicator (only for ISDN - V35/X21 modules).
- 7.- TCP/IP connection establishment indicator.
- 8.- Master module indicator.
- 9 and 10.- Input/output audio presence indicator, per channel.

Front module is the same for **ISDN - V35/X21** and **TCP/IP** communication boards (although firmware versions are different).

3.4.2. Back panel and connections description.



1 and 2.- ANALOG C1 and ANALOG C2 input and output connectors.

These connectors are “Hartman” type 4 Pin Connectors, two for Input and two for Output.

3 and 4.- DIGITAL I/O C1 and C2 digital input and output connectors.

These connectors are 9-Pin DB9 Female for the AES/EBU Inputs and Outputs. The Pin assignment for these connectors is:

<u>Pin no.</u>		<u>Connections</u>
1	-----	Rx N (V- input)
6	-----	Rx P (V+ input)
2	-----	Tx N (V- output)
7	-----	Tx P (V+ output)
5	-----	GND (ground)

- *When using the 64k modes, the AES/EBU signal from channel 1 will be in connector C1(mono signal) and AES/EBU signal from channel 2 will be in connector C2 (mono signal).*
- *When using the 128k (mono, stereo or dual), the AES/EBU signal will only be in C1(mono or stereo signal).*

5.- Auxiliary data channel. RS232 port.

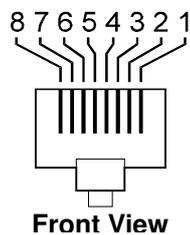
There is one DB-9 female chassis connector per module for auxiliary data transmission through the interface with an external computer and RS 232 protocol. A male-female cable should be used, and connected to your PC's port. The active contacts are:

	Course		PC
TX	2	----	2
RX	3	----	3
GND	5	----	5
RTS	8	----	8

Read more about datachannel at the end of this manual.

6.- RJ45 connector for IP connection.

This connector's pin arrangement is standard, and is shown as follows:



Cable number	Connections	
1	----- Tx	V+
2	----- Tx	V -
3	----- Rx	V+
4		
5		
6	----- Rx	V -
7		
8		

7.- Lock Screws for the Module

4. E@sy CONTROL SOFTWARE.

4.1. System description.

4.1.1. E@sy system.

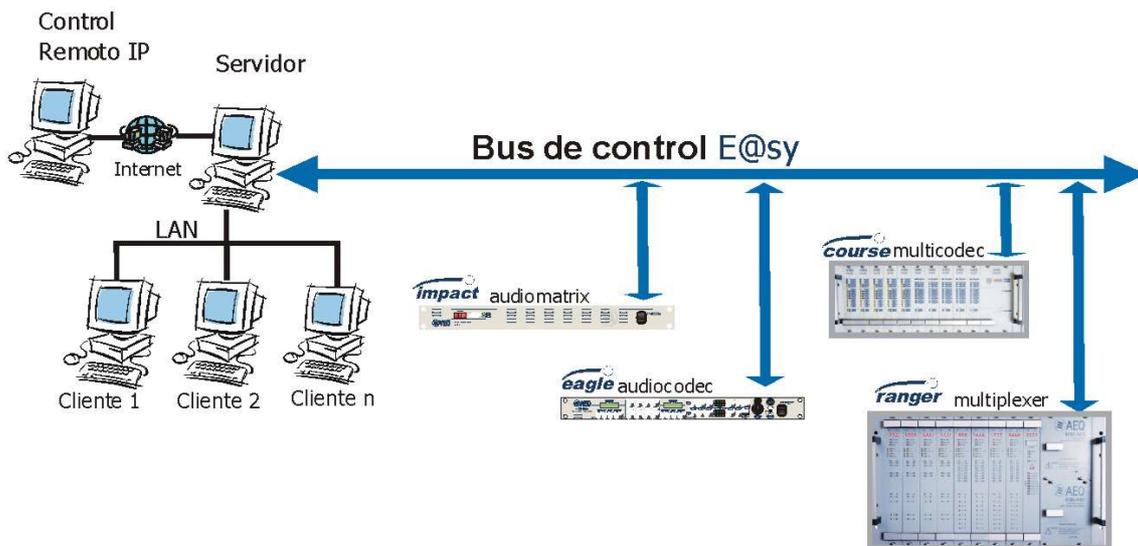
This equipment includes a new software technology called **E@sy**.

E@sy is an audio and communications digital devices integration technology that expands controlling capacity to a never before reached level and that includes both equipment and software applications. Equipments connect to a **server** computer through a RS-422 bus. Software applications in **client/server** computers form a high performance and powerful system.

The COURSE can be controlled remotely through the RS-422 bus. The basic **E@sy** software allows controlling from a single PC all the equipment connected to the same bus through a (LAN or WAN) IP network, setting encoding algorithms, calling, monitoring each line status, activating 'multiplex', line cut alarms, etc.

In all **E@sy** equipments, a firmware upgrading application for future upgrading is included. It will be explained in chapter 5 of this document.

E@sy connection diagram



4.2. Package description.

4.2.1. CD-ROM content.

In this package, you receive an auto-start up CD-ROM containing Manuals, the latest firmware files and this applications:

- **E@sy Server:** Server software application.
- **Course RTC:** Client/Control application.
- **Course FU:** Upgrading application.

To install the applications, insert the CD in your PC's CDROM drive and wait until the menu appears. In case it does not appear, check that the auto start feature of your CDROM drive is enabled.

4.2.2. Hardware and software requirements.

This system can work over an IP network or in a stand-alone PC.

In both cases, minimum requirements are:

- Compatible PC with a Pentium processor or higher, 32MB RAM Memory and a hard disk drive.
- When using a remote-control application, remote PC should have an IP connection with server PC. This can be through a LAN or a WAN network. Server computer must have one free serial port to connect with the COURSE through the RS-232 E@sy control port. In case of using the RS-422 E@sy control port, it will be necessary a RS232 to RS422 standard converter for interconnecting the PC and the COURSE.

Software runs on Windows NT, Windows 2000 and Windows XP operating systems.

This software requires the TCP/IP extension to be installed in the operating system. This extension is automatically installed when a network card or a modem is installed. In case of using a stand-alone PC, it is necessary to install the "MSLoopback" driver which is included in both operating systems installation disks.

E@syServer.exe program will be installed in the PC connected to the E@sy bus.

CourseRTC.exe will be installed in those remote computers from where equipment is going to be controlled. In case of a stand-alone computer, both programs should be installed in the same PC.

CourseFU.exe upgrading application works from a PC with direct connection to the COURSE audio codec.

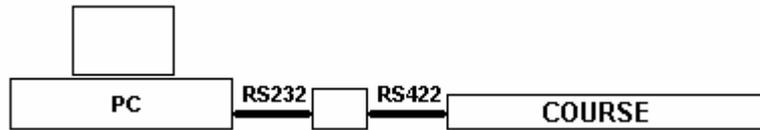
IMPORTANT NOTE: There is an incompatibility between the applications of E@sy system and processors with **Hyper Threading** or **Dual Core** options active. These options must be disabled through BIOS in order to avoid applications errors during its working.

4.3. Remote control system.

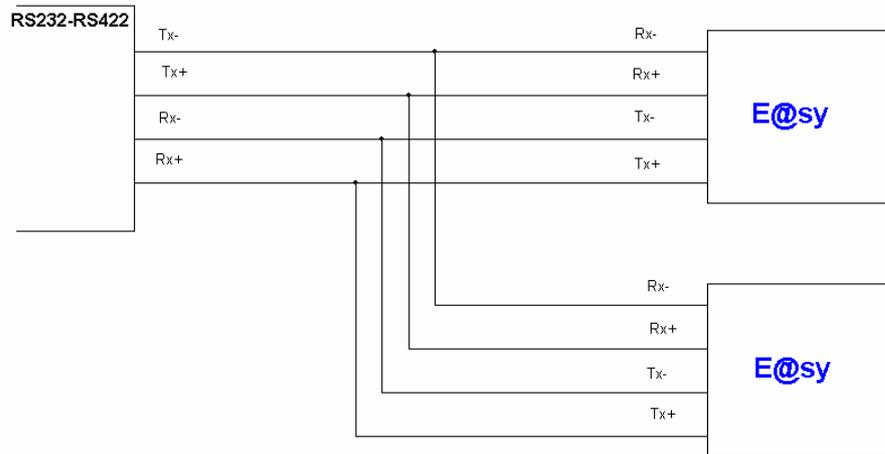
4.3.1. System connections.

The COURSE is controlled through one of the 2 control ports available: the RS-232 and the RS-422. The first one is a DB-9 female connector and the second one is a Hartmann type connector with 4 pins (2 for transmission and 2 for reception).

The COURSE will be connected to the server PC through a direct connection between PC serial port and RS-232 interface. This connections could also be made using the RS-422 interface by means of a RS-232/RS-422 standard converter.



If using an **E@sy** equipment network with several **E@sy** equipments, connection between those equipments should be made in parallel through RS-422 connector this way:



When all the connections are made, software has to be installed. If control application is going to be installed in a different PC than the server one, it is necessary to check that there is an IP connection between these computers. To do that, it is necessary to know the network name or IP address of the server PC inside the network. Once it's done, you'll just have to install the applications to start working.

Consult ANNEXE 1 in order to obtain more information about IP connections.

4.3.2. Server application.

E@syServer application is installed in the PC where the codecs are connected.

This application is the interface between the computer network and the control bus where the **E@sy** equipments are connected.

4.3.2.1. Installation.

Run the **E@syServer.EXE** application included on the CD and follow all the steps.

NOTES:

- Installing this software in a PC belonging to an intranet requires supervisor rights over the computer. In that case, consult your system supervisor.
- The first time this Install Shield Wizard is used, it will ask you to restart the system. If you have used this wizard before, It won't be necessary to restart.

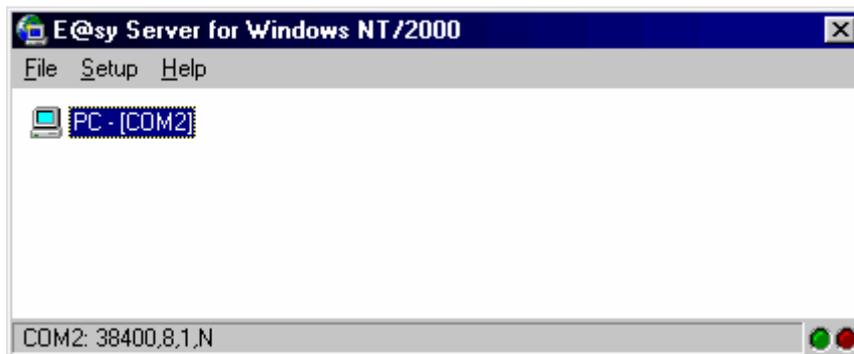
4.3.2.2. Operation.



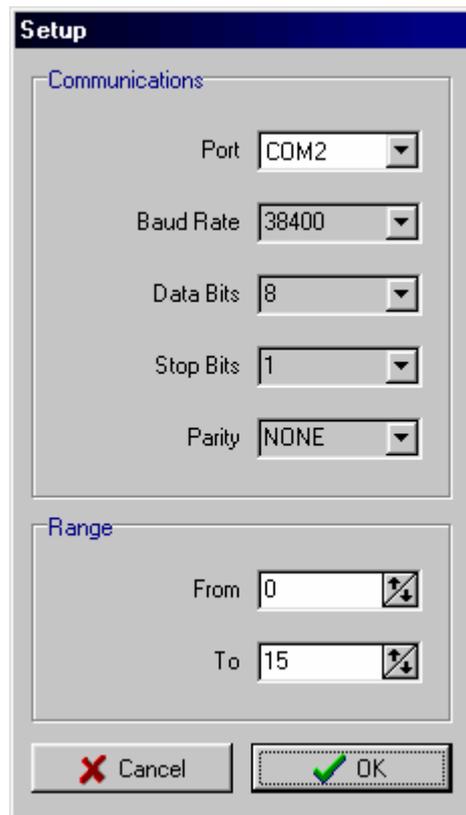
Once the installation is finished and the application is executed through the icon created in the desktop, the icon representing it will appear in the task bar:



Clicking over the icon with the mouse right key, '**SHOW**' option is showed, allowing to see the program window:



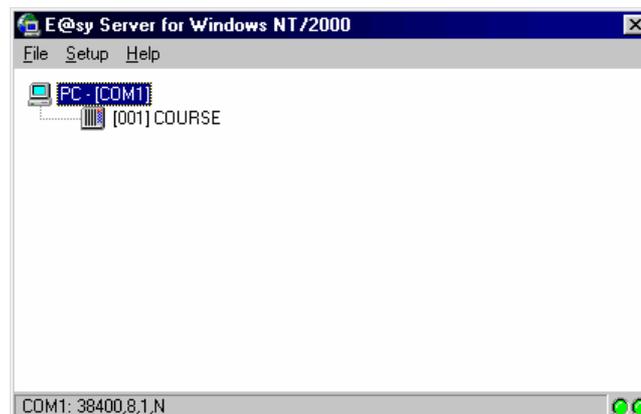
Select the **Setup** → **Port** option and configure in the **Communications** label the PC port to use.



E@sy equipment have a node number that can be configured to identify them in a network. Configure in the **Range** label the number of nodes that will be controlled, i.e., how many **E@sy** devices are available in the network. In order to optimize the communication, we recommend to establish a range as low as possible.

NOTE: Consult chapter 3.1.1 to know how to program this node number in this equipment.

Once all steps have been completed and all the connections have been made, program window will look like this:



In this case, there is a COURSE connected to the server PC identified as the 001 node and, from now on, it's possible to control it remotely.

This server program is resident and always active.

Setup → **S**ocket Server and **S**etup → **S**chedule options are used to control other **E@sy** equipments.

In the **F**ile menu, it is possible to search again the network devices with the **U**ppdate option and to close program window with the **C**lose option.

To stop server application and exit, it is necessary to click on the icon situated in the task bar with the right button and select the **E**xit option.

IMPORTANT: In order to avoid connection failures produced by accidental mistakes, the PC cannot be turned off until server application is stopped.

Uninstalling.

To uninstall this program, you should do it from the “Add or Remove Programs” panel of Windows clicking Start → Configuration → Control Panel.

In this menu, locate “**E@sy Server**”, click on it and choose the “Add or Remove” option to uninstall server software from the PC.

4.3.3. Client application.

4.3.3.1. Installation.

Run the “**CourseRTC.exe**” program which is contained in the CD-ROM and follow all the steps of the install shield wizard.

NOTES:

- Installing this software in a PC that is inside an intranet requires supervisor rights over the computer. In that case, consult your system supervisor.
- The first time this Install Shield Wizard is used, it will ask you to restart the system. If you have used this wizard before, it won't be necessary to restart.

4.3.3.2 Configuration.

When installation is finished, the **COURSE Real-time Control** program is executed by clicking on the icon created in the desktop.



IP Configuration

Before executing the program, IP configuration has to be made.

To configure it, the “**Course.INI**” file must to be edited. This file is in the path “C:\Program files\AEQ\E@sy Suite\” and will look like this:

```
[EASY SERVER]
PORT=60301
HOST=localhost
```

In ‘**HOST**’, the IP address or name of the server PC has to be set. By default, program is installed in local mode (HOST=localhost), that means that the COURSE is controlled from the same PC it is connected to. If this PC is not the same as the server one, this has to be changed to the new value.

DEMO Configuration

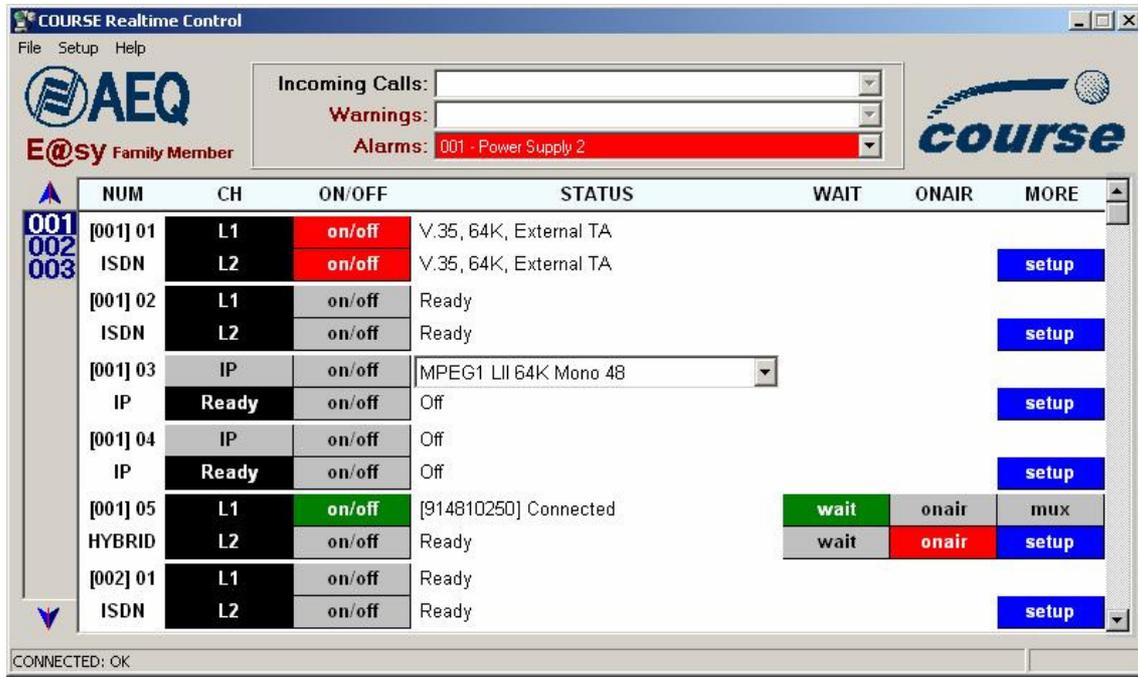
If the **E@sy Server** and **Course RTC** applications have been installed in a PC that is disconnected from the COURSE, for training or DEMO purposes, the file “**Course.INI**” should be modified through editing the line “**DEMO=0**”, changing this line for “**DEMO=1**”. If properly modified, the **Course Realtime Control** application will simulate a set of COURSE equipments on screen.

This changes on COURSE.INI file will be operative the next time the program is started.

4.3.3.3. Operation.

IMPORTANT NOTE: In order to configure and control the COURSE by means of **Course Realtime Control** application, it's necessary to run first **E@sy Server** application, that must always be active.

Once this has been done, the application may be started up, and it will look like this:



In this example, there are three COURSEs called '001', '002' and '003'. The first one has 5 boards (2 ISDN communication boards, 2 TCP/IP communication boards and 1 hybrid) and you can see the first board of the second COURSE (a ISDN communication board). The "NUM" column indicates the number of the node of each COURSE equipment, the slot number where each board is placed and the kind of board. The blue keys '001', '002' and '003' on the left part of the screen allow you to have a quick access to every COURSE in multiple equipment installations.

When there is an IP connection between the client computer and the COURSE, the following message will be displayed at the bottom of the screen "CONNECTED:OK".

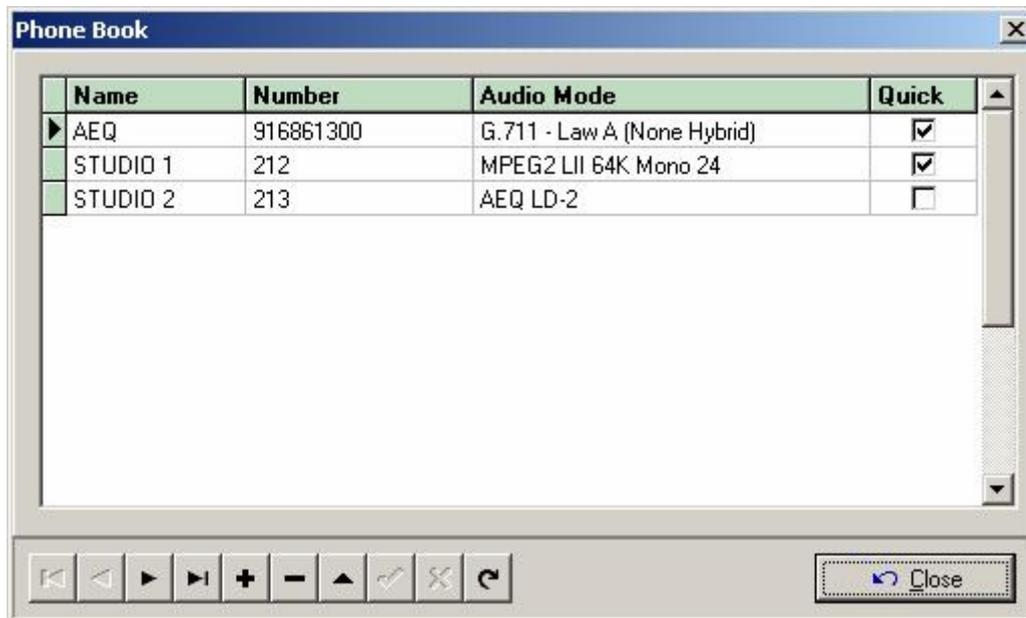
Otherwise, the following will be displayed "CONNECTION: REFUSED". If this appears, it will be necessary to review all the previous steps.

Once the system is in operation, the control panel for the equipment will be displayed, as well as a series of options in the Main Menu Bar.

4.3.3.3.1. Main Menu Bar.

The option **F**ile → **C**lose closes the control application.

The option **S**etup → **P**hone **B**ook shows the phone book for the equipment control software:



To enter a record in the phone book, you must type in the name and number directly over the fields and select the encoding mode. To enter a new record, just press the '+' button, and to delete one, use the '-' button.

For those records you would like to use from the quick access section in the dialing window, check the 'Quick' option. Thus, you will be able to store hundreds of numbers, but view only those that are going to be used at any given time.

The **S**etup → **I**P **P**hone option shows the IP addresses book for the equipment control software:



To enter a record in the IP book, you must type in the name, IP address and port directly over the fields. To enter a new record, just press the '+' button, and to delete one, use the '-' button.

For those records you would like to use from the quick access section in the dialing window, check the 'Quick' option. Thus, you will be able to store hundreds of addresses, but view only those that are going to be used at any given time.

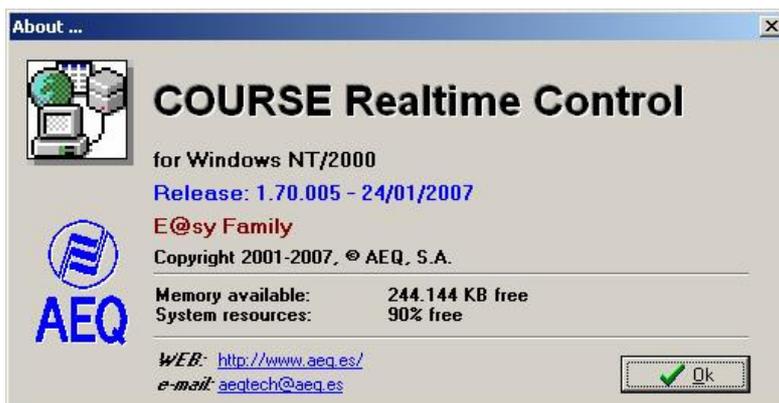
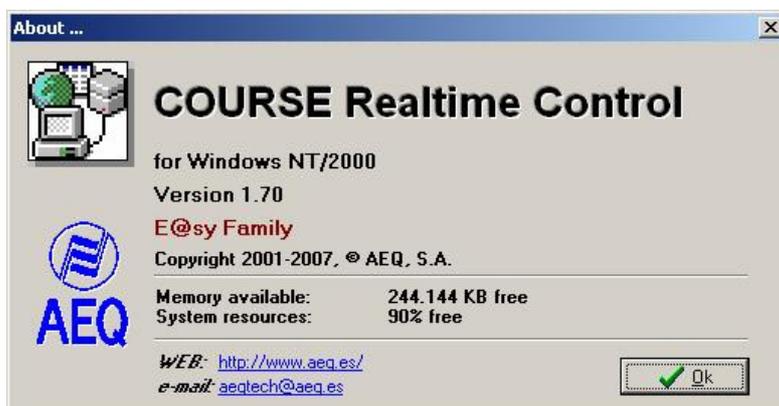
If the **Setup** → **Sirens (Warnings or Alarms)** option is activated, a sound simulating a siren will be heard when there is an error or failure in communications.

If **Setup** → **Ring (Incoming Calls)** is activated, it produces a sound simulating a telephone when there is an incoming call.

You must have a multimedia sound card installed to be able to hear these sounds.

The **Setup** → **Restore Window** option deactivates the Automatic Windows Maximize Feature (if they are minimized), that occurs when, for example, an incoming call appears.

The **Help** → **About** option indicates the installed Software Version. If double-clicked upon the AEQ logo, additional information about this Version appears. This information might be necessary in case that technical assistance should be required.



4.3.3.3.2. COURSE control panel.

The operation of the boards is controlled through this panel. The “NUM” column indicates the number of the node that each rack occupies, the slot number where each board is placed and the board type:

NUM	CH	ON/OFF	STATUS	WAIT	ONAIR	MORE
[001] 01 ISDN	L1	on/off	V.35, 64K, External TA			setup
	L2	on/off	V.35, 64K, External TA			
[001] 02 ISDN	L1	on/off	Ready			setup
	L2	on/off	Ready			
[001] 03 IP	IP	on/off	MPEG1 LIII 128K Mono 48			setup
	Ready	on/off	Off			
[001] 04 HYBRID	L1	on/off	[914810250] Connected	wait	onair	mux
	L2	on/off	Ready	wait	onair	setup

In this example, the panel shows that the COURSE equipment with node number ‘001’ has four boards: 2 ISDN communication boards (slots 1 and 2), 1 TCP/IP communication board (slot 3) and 1 hybrid (slot 4). It also shows the status of the two available lines in each board.

To see each line’s default encoding mode (for ISDN and TCP/IP boards), click on the ‘STATUS’ message in that line, and a drop-down menu will be displayed where default mode can be configured.

In this example, the line 1 of the TCP/IP communication board in slot 3 is configured with “MPEG1 LIII 128K Mono 48” mode and you can easily change this mode for another one just by selecting one of the options that are shown in the list:

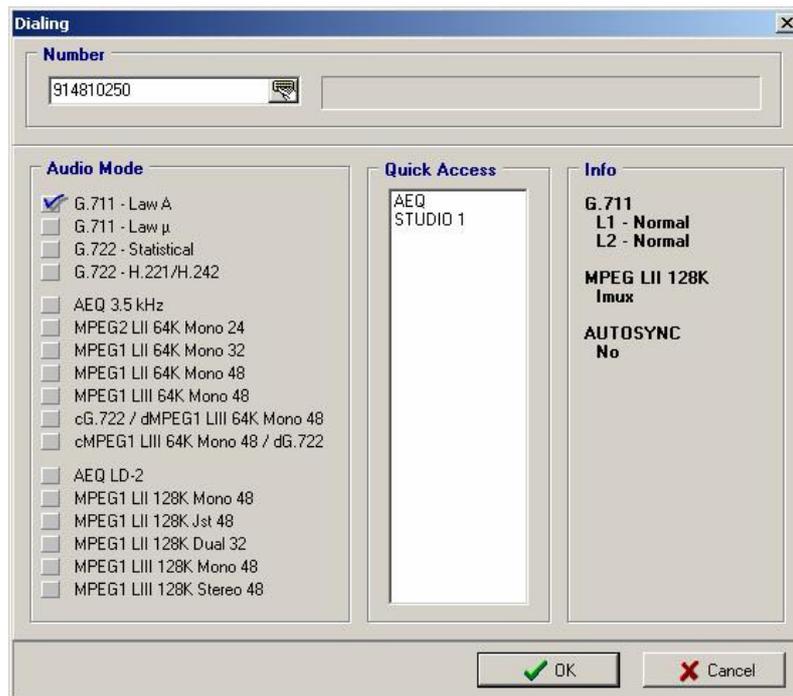
NUM	CH	ON/OFF	STATUS	WAIT	ONAIR	MORE
[001] 01 ISDN	L1	on/off	V.35, 64K, External TA			setup
	L2	on/off	V.35, 64K, External TA			
[001] 02 ISDN	L1	on/off	Ready			setup
	L2	on/off	Ready			
[001] 03 IP	IP	on/off	MPEG1 LIII 128K Mono 48			setup
	Ready	on/off	MPEG1 LII 64K Mono 48 MPEG1 LIII 64K Mono 48 AEQ LD-2 MPEG1 LII 128K Mono 48 MPEG1 LII 128K Dual 32 MPEG1 LIII 128K Mono 48 MPEG1 LIII 128K Stereo 48			
[001] 04 HYBRID	L1	on/off	[914810250] Connected	wait	onair	mux
	L2	on/off	Ready	wait	onair	setup

To **establish communication**, you must follow the following steps (depending on the kind of board):

- **ISDN - V35/X21 communication board.**

To establish communication you must press the **'on/off'** button corresponding to an active channel, with its GREY label. The RED label over 'on/off' button indicates that the dialing window will not appear for this particular channel, while a GREEN label indicates that the channel is active and connected, thus if you click upon the connection, it will be terminated.

If a GREY label **'on/off'** button, corresponding to a channel that is "ready", is clicked, the dialing window will be displayed:



The numbers that have been marked as **'Quick'** in the software phone book will be displayed in the **'Quick Access'** window and may be accessed by double clicking on them. It is also possible to change the audio encoding mode in the **'Audio Mode'** panel.

To call manually, you can type in the number in the **'Number'** window, or run the **'Speed Dial'** window by clicking on the key placed on the right:



It is also possible to change the audio encoding mode in the **'Audio Mode'** panel.

Once the parameters have been selected, press 'OK' and the equipment will make the call:

[001] 02	L1	on/off	[914810250] Calling ...
ISDN	L2	on/off	Ready

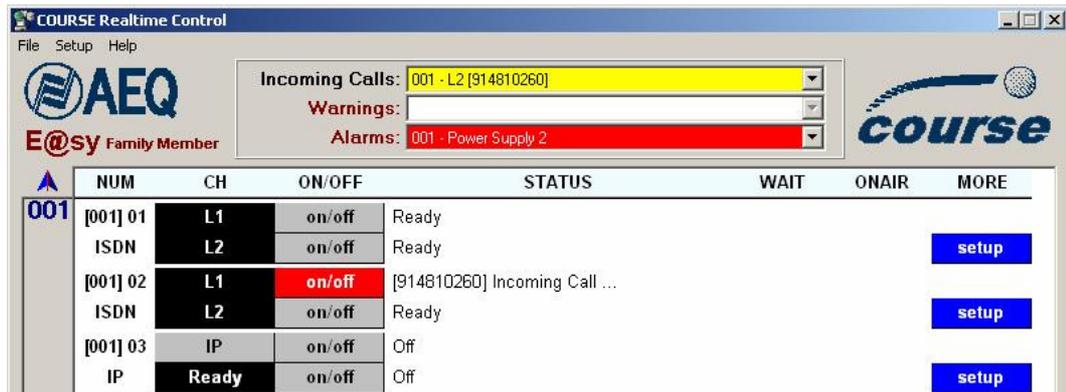
If the call is successful, a message will be displayed indicating that a connection has been established and the used audio encoding mode (it can be easily changed for another one just by clicking on the message and selecting one of the options that are shown in the drop-down menu):

[001] 02	L1	on/off	[914810250] G.711 - Law A
ISDN	L2	on/off	Ready

Otherwise, if there is any problem with the ISDN line, an error message is displayed, such as:

[001] 02	L1	on/off	ISDN LINK ERROR
ISDN	L2	on/off	Ready

When a call is received, the equipment will show the message 'Incoming Call' on the corresponding board line and this event is also shown at the top of the window ('Incoming Calls'):



The screenshot shows the 'COURSE Realtime Control' window. At the top, there are dropdown menus for 'Incoming Calls' (set to '001 - L2 [914810260]'), 'Warnings', and 'Alarms' (set to '001 - Power Supply 2'). Below this is a table with columns: NUM, CH, ON/OFF, STATUS, WAIT, ONAIR, and MORE. The table shows three channels:

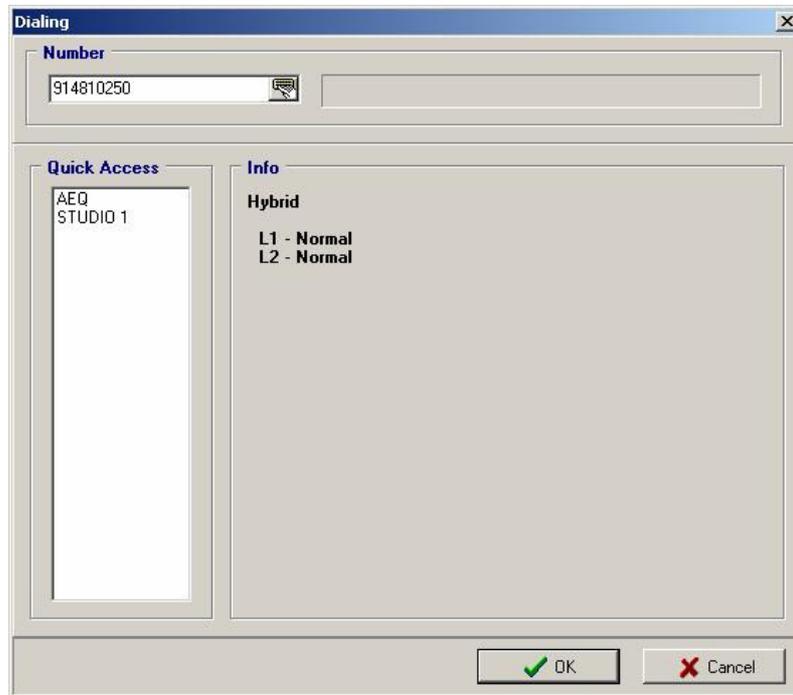
NUM	CH	ON/OFF	STATUS	WAIT	ONAIR	MORE
001	[001] 01	L1	on/off	Ready		
	ISDN	L2	on/off	Ready		setup
001	[001] 02	L1	on/off	[914810260] Incoming Call ...		
	ISDN	L2	on/off	Ready		setup
001	[001] 03	IP	on/off	Off		
	IP	Ready	on/off	Off		setup

To connect the call, you must press the 'on/off' button in the channel that is receiving the call, unless that channel is configured as **automatic Off-Hook**.

- **Hybrid board.**

To establish communication you must press the **'on/off'** button corresponding to an active channel, with its GREY label. A GREEN label indicates that the channel is active and connected, thus if you click upon the connection, it will be terminated.

If a GREY label **'on/off'** button, corresponding to a channel that is "ready", is clicked, the dialing window will be displayed:



The numbers that have been marked as **'Quick'** in the software phone book will be displayed in the **'Quick Access'** window and may be accessed by double clicking on them.

To call manually, you can type in the number in the **'Number'** window, or run the **'Speed Dial'** window by clicking on the key placed on the right:



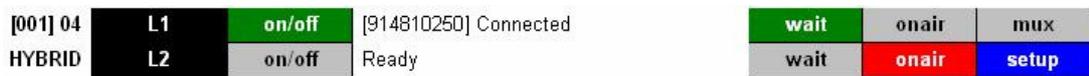
Once the number has been selected, press 'OK' and the equipment will make the call (a message will be displayed indicating that a connection has been established):



When a call is received, the 'on/off' button start blinking on red and 'Incoming Call...' message appears. Press the button to establish connection.



When a connection has been established, each channel can be set on **Wait** or **On Air** mode or **Multiplex** mode can be activated by pressing the corresponding button:



On **Wait** mode, the other extreme of the line listen to the program but it doesn't get in. If the button is pressed again, connection is still established but no audio is sent.

On **On Air** mode, the other extreme of the line listen to the program and get in. If the button is pressed again, connection is still established but no audio is sent or received.

Wait and **On Air** modes are mutually exclusive options. When a channel is on **Wait** mode and you activate **On Air** mode, the first mode deactivates and vice versa.

Regarding **Multiplex** mode, if both lines are connected and **On Air**, it's possible to establish a full duplex communication between these two lines and the hybrid board, establishing a multiconference (each line is able to send audio to the other two ones and receive audio from them).

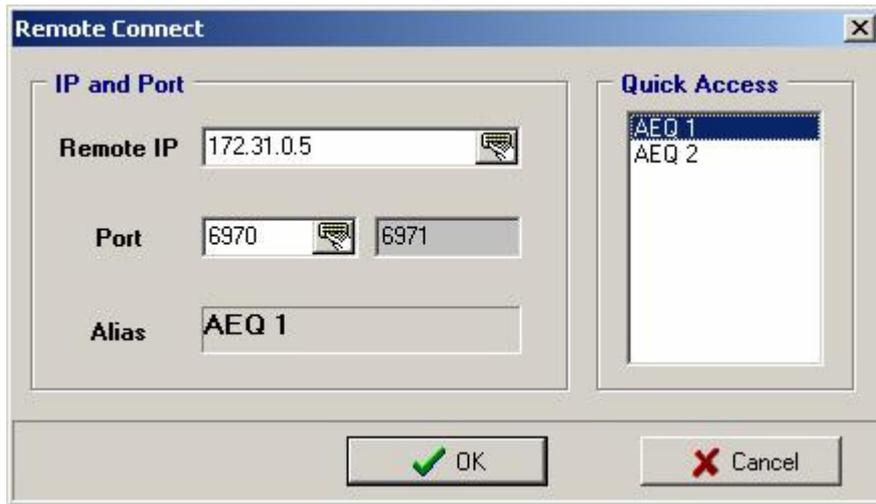
To terminate a connection, press the 'on/off' button and a confirmation window will appear:



- **TCP/IP communication board.**

To establish communication you must press the 'IP' button corresponding to an active channel, with its GREY label. The RED label indicates that the dialing window will not appear ('Link Error' message appears). The ORANGE label indicates that communication is being established and a GREEN label indicates that communication is yet established.

If a 'IP' button, corresponding to a board that is "ready", is clicked, the dialing window will be displayed:



The addresses that have been marked as 'Quick' in the software IP addresses book will be displayed in the 'Quick Access' window and may be accessed by double clicking on them (in 'Alias' window the name of selected address will appear).

To call manually, you can type in the address in the 'Remote IP' window, or run the 'Speed Dial' window by clicking on the key placed on the right:



Once the IP address has been selected, press 'OK' and the equipment will make the call ('Connecting...' indication and destination IP address or name will be shown alternately):

[001] 03	IP	on/off	Off	[001] 03	IP	on/off	Off
IP	Connecting ..	on/off	Off	IP	AEQ 1	on/off	Off

If the call is successful, a message will be displayed indicating that a connection has been established ('**Connected**' indication and destination IP address or name will be shown alternately):

[001] 03	IP	on/off	Off	[001] 03	IP	on/off	Off
IP	Connected	on/off	Off	IP	AEQ 1	on/off	Off

In a TCP/IP communication one of the extremes will always work as master and the other one as slave. Master sets encoding and synchronization. Slave follows automatically the encoding changes and recovers synchronization.

Once the TCP/IP communication is established, you can active audio connection independently per channel. When you press the '**on/off**' button of an available channel (grey label), '**Connected**' indication and encoding mode will appear:

[001] 03	IP	on/off	Connected MPEG1 LIII 128K Mono 48
IP	Connected	on/off	Off

Encoding mode can be easily changed for another one just by clicking on that indication (on the board that is working as master) and selecting one of the options that are shown in the drop-down menu:

[001] 03	IP	on/off	MPEG1 LIII 128K Mono 48
IP	AEQ 1	on/off	<ul style="list-style-type: none"> MPEG1 LII 64K Mono 48 MPEG1 LIII 64K Mono 48 AEQ LD-2 MPEG1 LII 128K Mono 48 MPEG1 LII 128K Jst 48 MPEG1 LII 128K Dual 32 MPEG1 LIII 128K Mono 48 MPEG1 LIII 128K Stereo 48

When you press a channel '**on/off**' button and there is no TCP/IP communication established, the following message will appear:



When TCP/IP communication is established and you active the audio connection in one channel of master module, '**Incoming Audio...**' message will appear in the same channel of slave module:

[001] 04	IP	on/off	Incoming Audio...
IP	Connected	on/off	Off

To activate that connection, you must press the '**on/off**' button in that channel, unless the '**Audio Slave Connection**' of that channel is configured as **automatic**. Once reception is active, channel indicates connected and also shows encoding mode (it's set by the master module).

To terminate audio connection, press the **'on/off'** button of the corresponding channel (only in master module when **'Audio Slave Connection'** is configured as automatic or in both modules when it's configured as manual).

If there is any problem with the IP connection, an error message is displayed, such as:

[001] 03	IP	on/off	Off
IP	Link Error	on/off	Off

To terminate an IP connection, press the **'on/off'** button of master module and a confirmation window will appear:

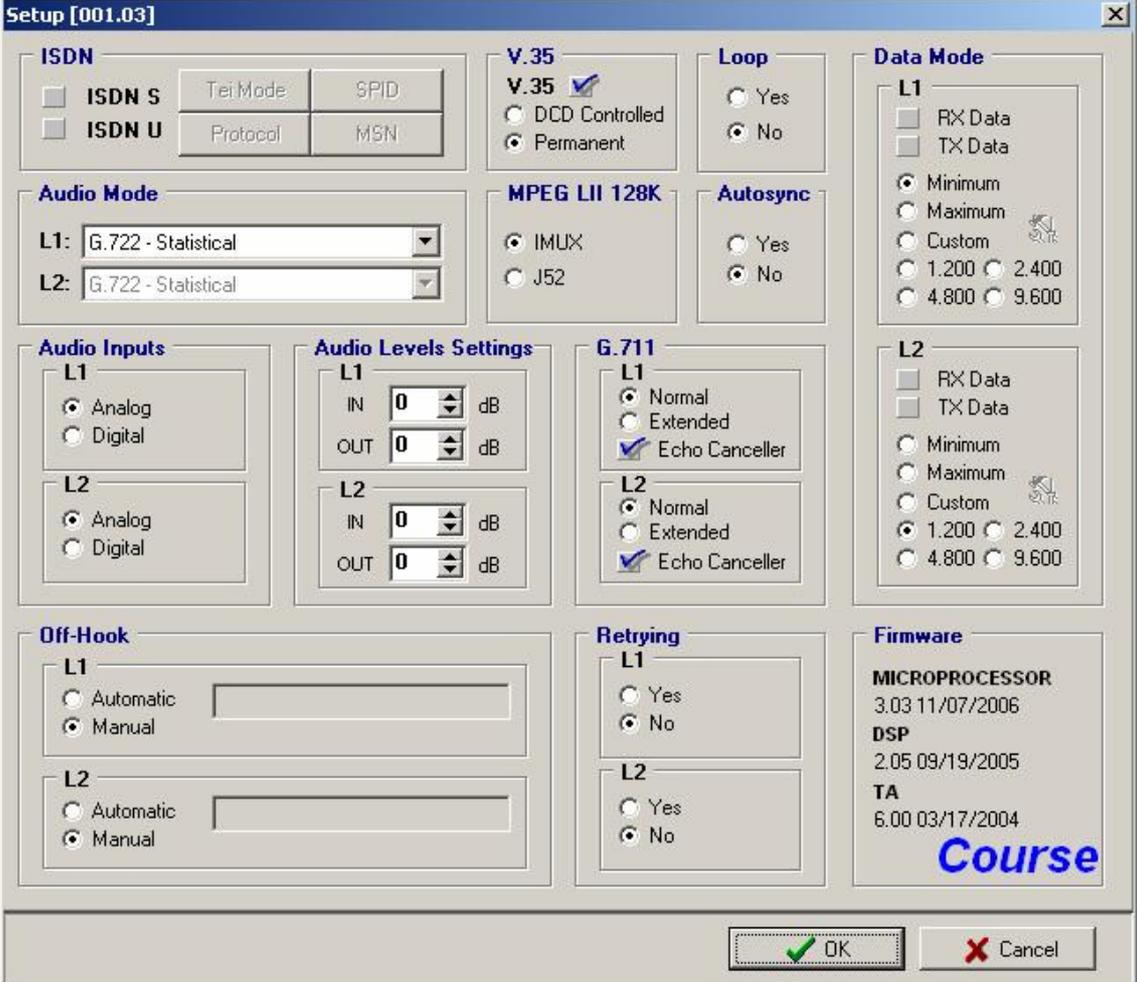


4.3.3.3.3. Modules configuration window.

The configuration window of the selected module is accessed by clicking on the blue 'setup' button placed in the "MORE" column.

4.3.3.3.3.1. ISDN - V35/X21 modules configuration.

The configuration window will look like this:



The configurable parameters are:

- **Type of connection:** Allows you to select the type of connection for each individual module to the Digital Network. The possible types are ISDN S, ISDN U y V35.
 - **ISDN S:** Is used for Basic Rate Interface (BRI) ISDN with EURO-ISDN (ETSI) Protocol. These connections are physically of 4-Wire type and the Service Provider is also providing the Network Terminal.
 - **ISDN U:** Is used for Basic Rate Interface (BRI) ISDN with ANSI Protocol. These connections are physically of 2-Wire type, so the Network Terminal is inside the Proper Equipment and NOT provided by the Service Provider.
 - **V.35:** Is used for permanent connections through the V.35 Port or through an External Terminal Adapter (TA). The Bit Rate of the External Terminal Adapter is automatically detected by the Course if the TA is generating a clock at 64kbps, 128 kbps or 256 kbps.
It can be controlled either through a Permanent Stream or through DCD (Data Carrier Detect).

In case of **ISDN** connection the following parameters must be configured:

- **Protocol:** Defines the protocol that is used by the Terminal Adaptor. There are two options; NATIONAL-1, the most commonly used in the United States, and the EUROISDN, that is the Standard Protocol for ISDN in Europe and widely used all over the world.



- **Tei Mode:** Allows to configure the mode of assignation of the TEI Identifier to either DYNAMIC or FIXED Mode. In Dynamic Mode it allows the Network to assign the TEI and in Fixed mode the TA forces the TEI=0. Generally, this configuration should be set to DYNAMIC mode.



- **SPID (Service Profile Identification Numbers):** Whenever using the NATIONAL-1 Protocol, it is necessary to configure the SPID's (Service Profile Identification Number) for each TA. These numbers are assigned indicated by the Service Provider or each B-channel of the ISDN Line. Without the SPID's correctly configured, the Terminal Adaptor will not function.



- **MSN (Multiple Subscriber Number):** This is used in EUROISDN. If these numbers are indicated it is possible to select what B-channel answers the incoming calls.



- **Audio Mode:** Allows to select the default Audio Encoding mode for each line (128 kbps modes) or B-Channel (64kbps Modes).
- **Audio Inputs:** Allows to select Analogue or Digital (AES/EBU) Audio Inputs.
- **Audio Level Settings:** Allows to configure input and output gain for each line between -12 and +12 dB (when encoding mode is MPEG LIII output gain is disable).
- **G.711:** This allows activating the Frequency extender mode in G.711 whenever used in a remote connection to an AEQ Telephone Hybrid that incorporates this Function.

Typical phone circuits have a limited bandwidth of between 300 and 4000 Hz. Unfortunately, most voice energy is in frequencies lower than 300 Hz, which are lost during transmission through the phone line. For this reason, the audio signal has the typical phone sound, with the loss of body in the voice being, perhaps, its most representative aspect.

In addition to hybrid boards, **AEQ** has equipment able to transmit signals between 50 and 3750 Hz. This mode of operation is called “**extended mode**”. Evidently, this mode allows the band of frequencies included between 50 and 300 Hz to be transmitted through the phone line. To achieve this, the signal sent undergoes a 250 Hz frequency shift, improving the quality of the signal received in its lower frequencies at the expense of the higher band. The bandwidth transmitted is, thus, between 50 and 3750 Hz. The 250 Hz lost from the higher frequencies are not very significant, given the logarithmic nature of the audio frequency response.

The translation of frequencies takes place by encoding the audio signal before it is sent to the phone line. Decoding, i.e., the reverse frequency shift, takes place at the receiving end; this must be done by the appropriate equipment. The decoded signal has the original bandwidth (50-3750 Hz) without having suffered any alteration. Thus, greater depth and clarity is achieved in the voice signal, even in those communications that take place under the worst conditions.

- **G.711 Extended:** To establish communications (in both directions) on non-digital lines (basic telephone network), as long as there is an hybrid board or an AEQ TLE02, TLE02D, TH-02 MkII, TH-02 MkIII or MPAC02 type equipment at the other end of the line, with the same extended frequency function activated.
- **G.711 Normal:** In all other cases.

It is possible to activate an **automatic remote echo canceller**, that can improve audio quality of connections with analog telephone lines.

- **Retrying:** Allows selecting the retrying mode whenever the connection has not been successful. The equipment will redial as many times as necessary in order to establish the desired connection.
- **Autosync:** With this option activated the equipment will go into SLAVE Mode and search the encoding algorithm that the remote equipment is programmed to use for transmission. If the remote equipment should change the Encoding Algorithm during the transmission the Course will re-synchronize to this new mode automatically.
- **MPEG LII 128K:** This option is to establish the Framing Mode for 128 kbps modes in J.52 or IMUX.

The normal operating mode is the one that uses J.52 type multiplexing; however, for the EAGLE to be compatible with TELOS (ZEPHYR model) and CCS (CDQ model) audio codecs for 128 kbps modes, it is necessary to select IMUX

- **Loop:** This activates the LOOP Mode function. This consists in that the equipment is returning the through its Output the signal that is present at its Input at the same time as an ISDN LOOP is generated. This means that it is also returning to the remotely connected equipment its proper generated and transmitted signal.
- **Off-Hook:** The equipment will Automatically Off-Hook the line or not whenever receiving a call. If in the Associated Numeric Field is introduced an ISDN Number, the equipment will only OFF-Hook calls from this Number.
- **Data mode:** It is possible to enable one transparent two-way end-to-end data channel along with each audio channel, to which we can interface data terminal equipment between both units (for example, computers).

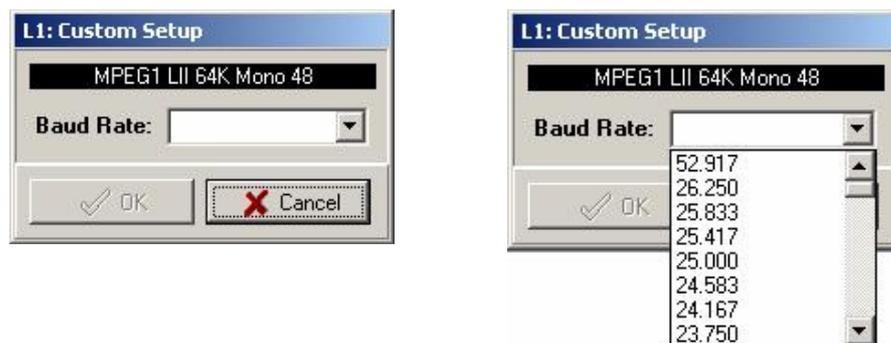
This is a transparent two-way variable-speed end-to-end link, one for each audio channel.

The data channel may be set up according to the encoding mode being used in the connection that is established. Data channels are not possible in G. 711 and G. 722 STAT voice modes.

There are different possible connection configurations:

- **Maximum:** where you can transmit and receive at up to 64 kbps in one 64 kbps connection (one B channel), or transmit and receive at up to 115 kbps in a 128k connection. With this option, the audio bandwidth may be reduced if the data flow is large, and an interruption in the audio may occur because of the data transmission requirements. Physical connection to the RS232 port has to be done at 115200bps.
- **Minimum:** where the speed is set at 1200bps. In this option, the quality of the audio being transmitted does not deteriorate. Physical connection to the RS232 port has to be done at 115200bps.
- **Custom:** where the transmission speed is defined by the user according to the application requirements. Physical connection to the RS232 port has to be done at 115200bps.

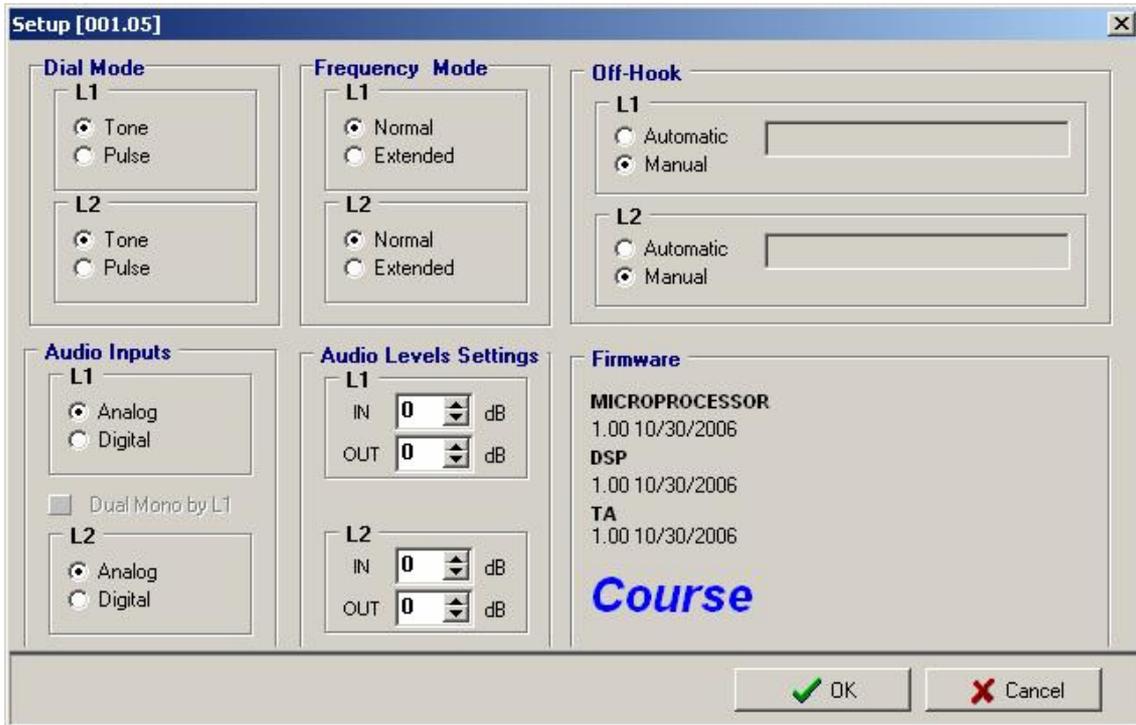
The “Custom Setup” window allows for the configuration of these Binary Rates:



- **1200/2400/4800/9600**, that make the port to work physicaly at the selected baudrate. In this case, physical connection to the RS232 is done at the selected baudrate.
- **Firmware:** It shows the Firmware Versions that are installed on the software modules of the selected board.

4.3.3.3.2. Hybrid modules configuration.

The configuration window will look like this:



The configurable parameters are:

- **Dial Mode:** Allows selecting **tone** or **pulse** dialing mode.
- **Frequency Mode:** This allows activating the Frequency extender mode.

Typical phone circuits have a limited bandwidth of between 300 and 4000 Hz. Unfortunately, most voice energy is in frequencies lower than 300 Hz, which are lost during transmission through the phone line. For this reason, the audio signal has the typical phone sound, with the loss of body in the voice being, perhaps, its most representative aspect.

In addition to hybrid boards, **AEQ** has other equipments able to transmit signals between 50 and 3750 Hz. This mode of operation is called “**extended mode**”. Evidently, this mode allows the band of frequencies included between 50 and 300 Hz to be transmitted through the phone line. To achieve this, the signal sent undergoes a 250 Hz frequency shift, improving the quality of the signal received in its lower frequencies at the expense of the higher band. The bandwidth transmitted is, thus, between 50 and 3750 Hz. The 250 Hz lost from the higher frequencies are not very significant, given the logarithmic nature of the audio frequency response.

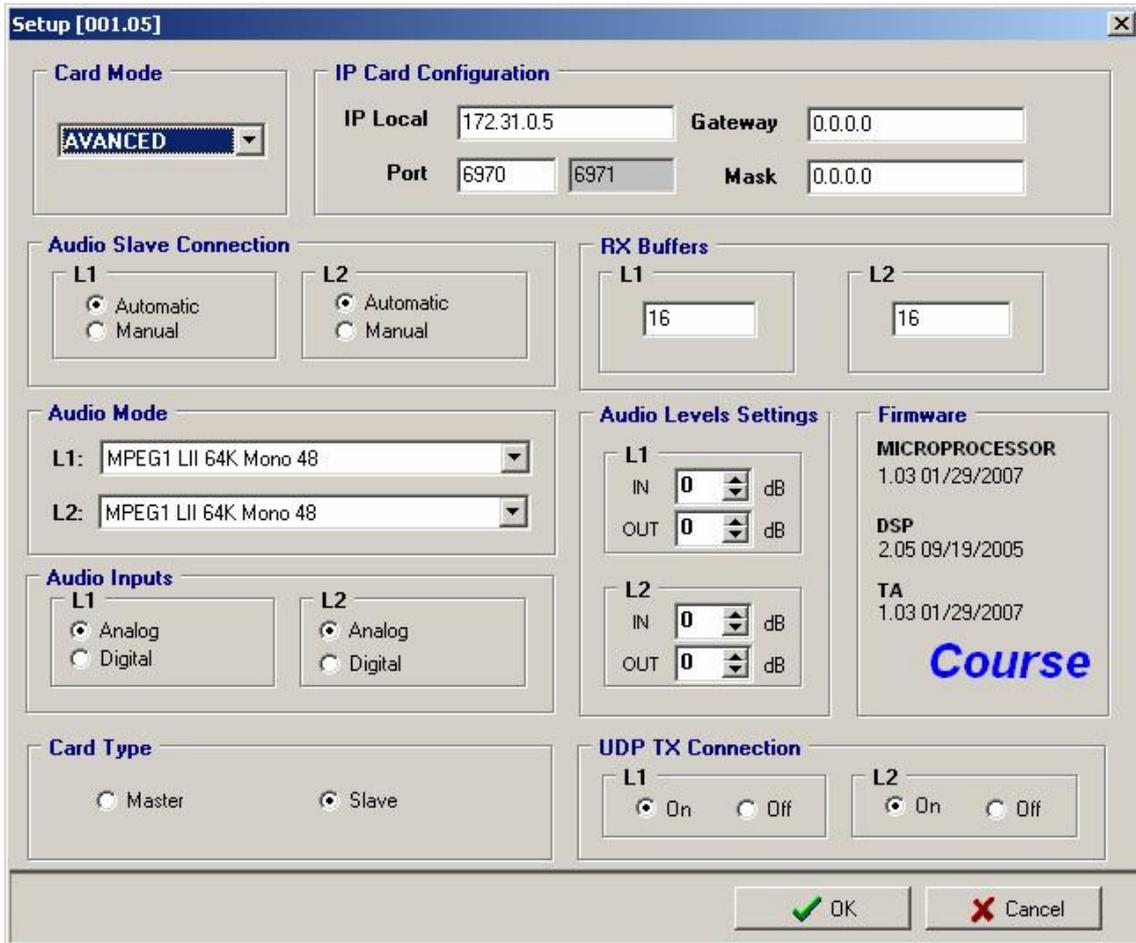
The translation of frequencies takes place by encoding the audio signal before it is sent to the phone line. Decoding, i.e., the reverse frequency shift, takes place at the receiving end; this must be done by the appropriate equipment. The decoded signal has the original bandwidth (50-3750 Hz) without having suffered any alteration. Thus, greater depth and clarity is achieved in the voice signal, even in those communications that take place under the worst conditions.

- **Off-Hook:** The equipment will Automatically Off-Hook the line or not whenever receiving a call. If in the Associated Numeric Field is introduced a phone number, the equipment will only Off-Hook calls from this Number.

- **Audio Inputs**: Allows to select Analogue or Digital (AES/EBU) Audio Inputs.
- **Audio Level Settings**: Allows to configure input and output gain for each line between -12 and +12 dB.
- **Firmware**: It shows the Firmware Versions that are installed on the software modules of the selected board.

4.3.3.3.3. TCP/IP modules configuration.

The configuration window will look like this:



The configurable parameters are:

- **Card Mode:** Allows you to select the board working mode. The available options are:
 - **NORMAL:** In this mode, the system will work as a conventional telephone system, and the use will be easier for users. “**Card Type**” and “**UDP TX Connection**” options will be disabled and automatically controlled depending on the system status (no calls, incoming call, outgoing call, etc).
 - **MULTICAST TX:** In **Multicast** mode, there is a transmitter and several receivers. Both transmitter and receivers must connect to the remote address of a multicast group. In **MULTICAST TX** mode the board is configured as transmitter.
 - **MULTICAST RX:** In this mode, the board is configured as receiver in **Multicast** mode.
 - **ADVANCED:** In this mode, user can configure all the connection parameters.

- **IP Card Configuration:** Allows you to configure the following parameters:
 - **IP Local:** This is the IP address of the board. It consists of four numbers separated by dots: the value of each of these numbers ranges from 0 to 255. The system works just like any other IP device, with no limits about it (except the characteristic ones of the network configuration).

IMPORTANT NOTE: In **Multicast** mode an IP address of suitable range (from 224.0.0.0 to 239.255.255.255) must be used. Please notice that there are some reserved addresses (IANA: Internet Assigned Numbers Authority).
 - **Port:** This is the local port. The network administrator must configure the router to forward the UDP ports used in the communication: by default this port is **6790** for RTP and **6791** for RTCP.
 - **Gateway:** This is the access gate to an external network. So that the system can be accessed from an external network, you will need to establish a suitable gateway address. This address should be established by the administrator of the network. By default this address is 0.0.0.0.
 - **Mask:** Sets up a subnet mask to establish the minimum required matching between the IP addresses of the diverse system elements that will enable them to communicate with each other. For example, with a mask of the 255.255.0.0 type, all the system components that have IP addresses of the 172.31.X.X type will be able to communicate with each other. If one or more of the first numbers of the IP addresses does not match up with the rest of the elements, the subnet mask will prevent it from communicating with the other system components.
- **Audio Slave Connection:** Once the TCP/IP link is established, the audio transmission/reception can be controlled independently per channel. Regarding the slave, the connection for each channel can be set **manual** or **automatic** (when master asks for it).
- **RX Buffers:** Allows to establish the reception buffer size. This buffer makes up for the jitter caused by the network, by inserting a constant delay in audio. Its size is an adjustable parameter that will be adjusted based on network quality.
- **Audio Mode:** Allows to select the default Audio Encoding mode for each line.
- **Audio Inputs:** Allows to select Analogue or Digital (AES/EBU) Audio Inputs.
- **Audio Level Settings:** Allows to configure input and output gain for each line between -12 and +12 dB (when encoding mode is MPEG LIII output gain is disable).
- **Card Type:** Allows to select if the board will work as **Master** or **Slave**. In a TCP/IP communication one of the extremes will always work as master and the other one as slave. Master sets encoding and synchronization. Slave follows automatically the encoding changes and recovers synchronization.
- **UDP TX Connection:** Allows to enable (**On**) or not (**Off**) the audio transmission for each channel independently (audio reception is always enable).
- **Firmware:** It shows the Firmware Versions that are installed on the software modules of the selected board.

5. FIRMWARE UPGRADE.

WHEN DO YOU NEED TO UPGRADE?

The COURSE leaves the factory with the latest available versions at that time. However, it is possible that later versions with new capabilities will appear in the future; these will be published at an opportune time on AEQ's website:

<http://www.aeq.es>

In case of doubt, contact the Technical Assistance Service:

sat@aeq.es

PRECAUTIONS BEFORE UPGRADING:

It is advisable to disconnect the established connections to upgrade the equipment.

5.1. System connections.

To upgrade the equipment's firmware, the COURSE will connect to a PC through a connection between one of the PC's serial ports and the RS-232 **E@sy** control port. In case of using the RS-422 **E@sy** control port, it will be necessary a RS-232 to RS-422 standard converter to make this connection.

If there are several COURSE units connected in parallel to the RS-422 bus, the upgrading software will show the versions of each of them.

5.2. Installation of the upgrading software.

By running the "**CourseFU.exe**" file, you start up installation.

You must follow all the steps indicated by the installation wizard, until the process is complete.

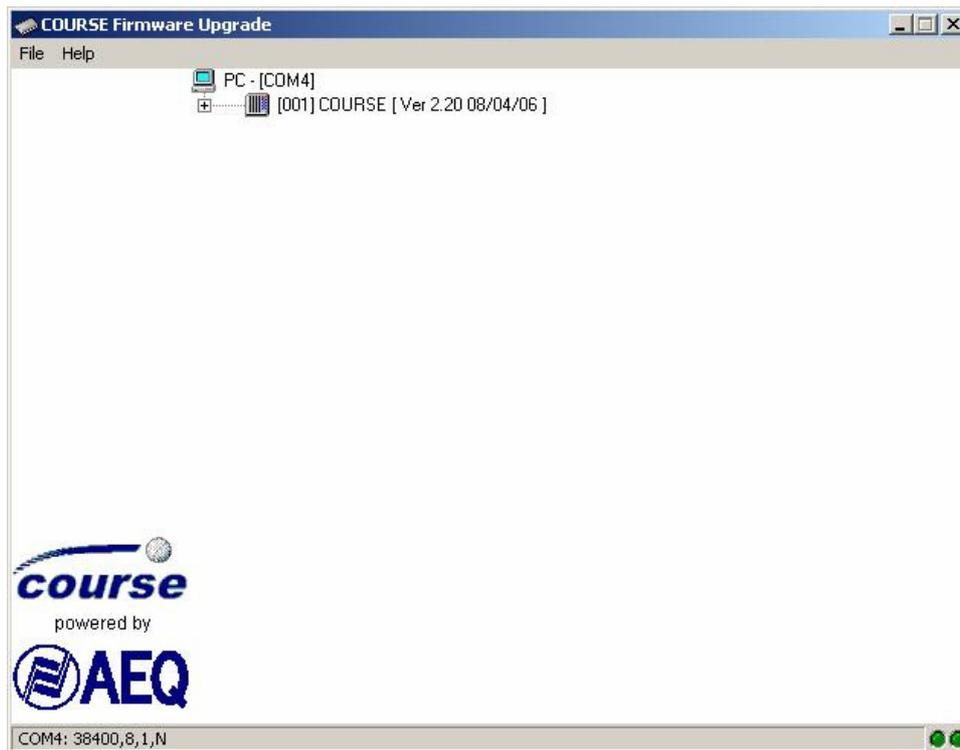
5.3. Operation.

IMPORTANT NOTE: Before running the upgrading software, you must close **Course Realtime Control** and **E@sy Server** applications.

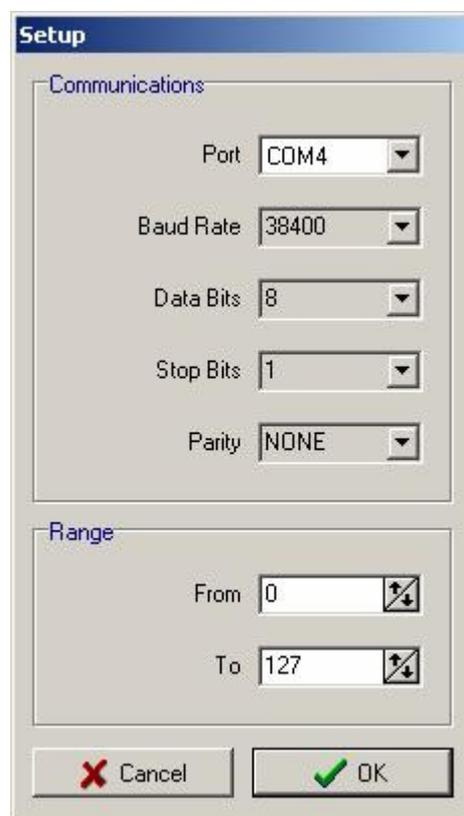
Double click on the icon to run the application:



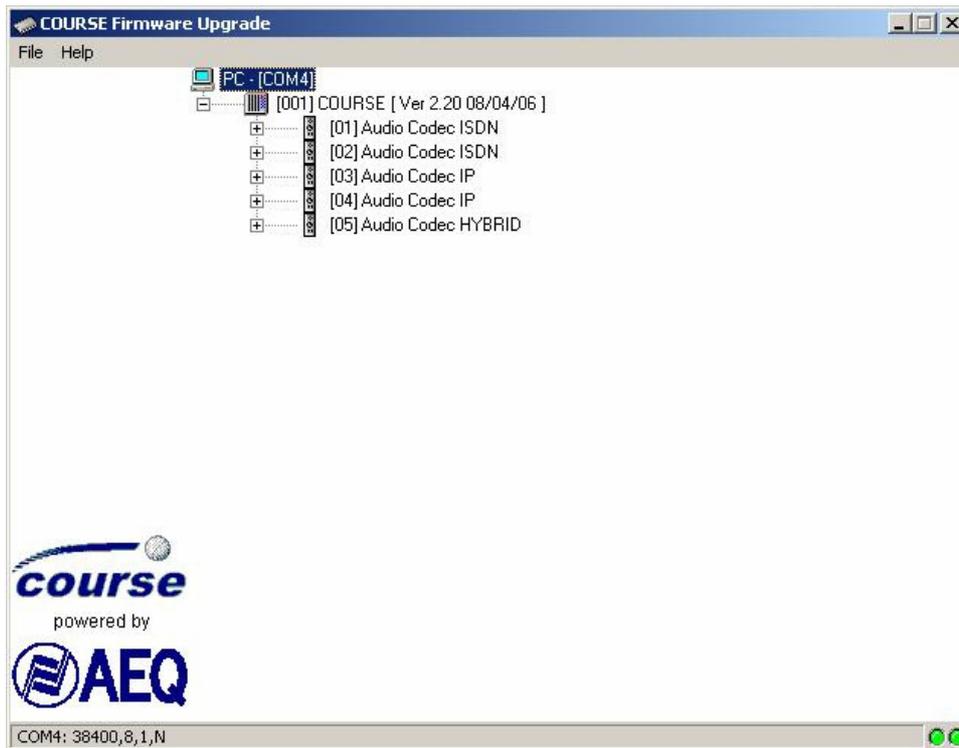
Once the application has run, the program window will be displayed:



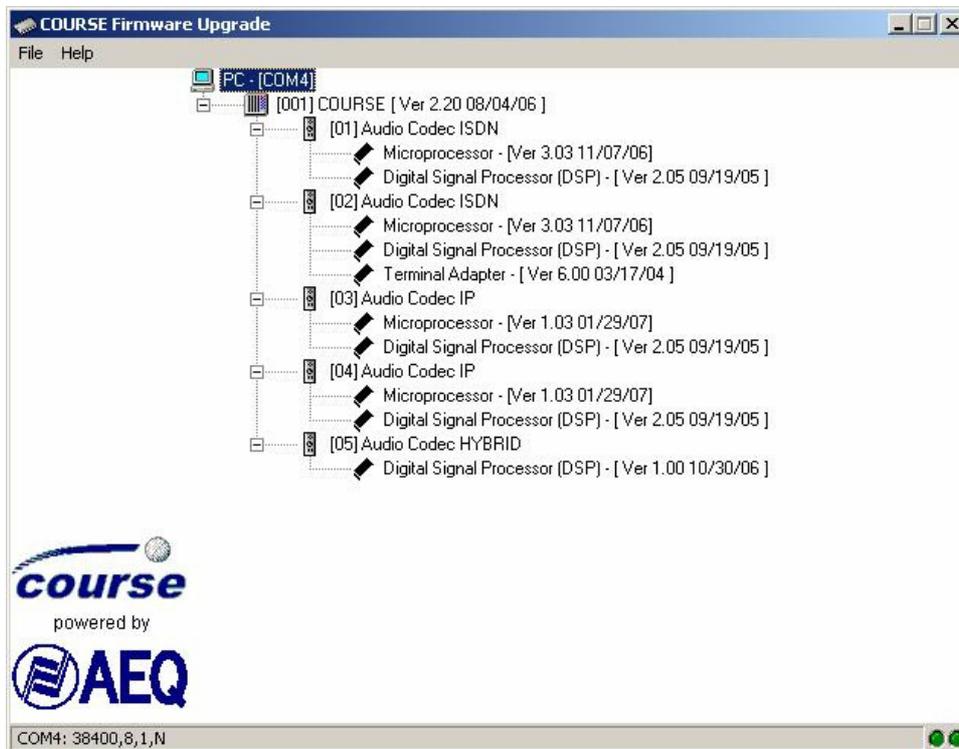
In the **File** → **Setup PC** window, you establish the **port** of the PC that will be connected with the COURSE and the **range** of node numbers to do the equipments search (the node number identify the **E@sy** equipments in a network). In order to optimize the communication, we recommend to establish a range as low as possible.



By clicking on the '+' sign of COURSE unit, the structure is expanded, allowing you to see the boards of the connected unit:



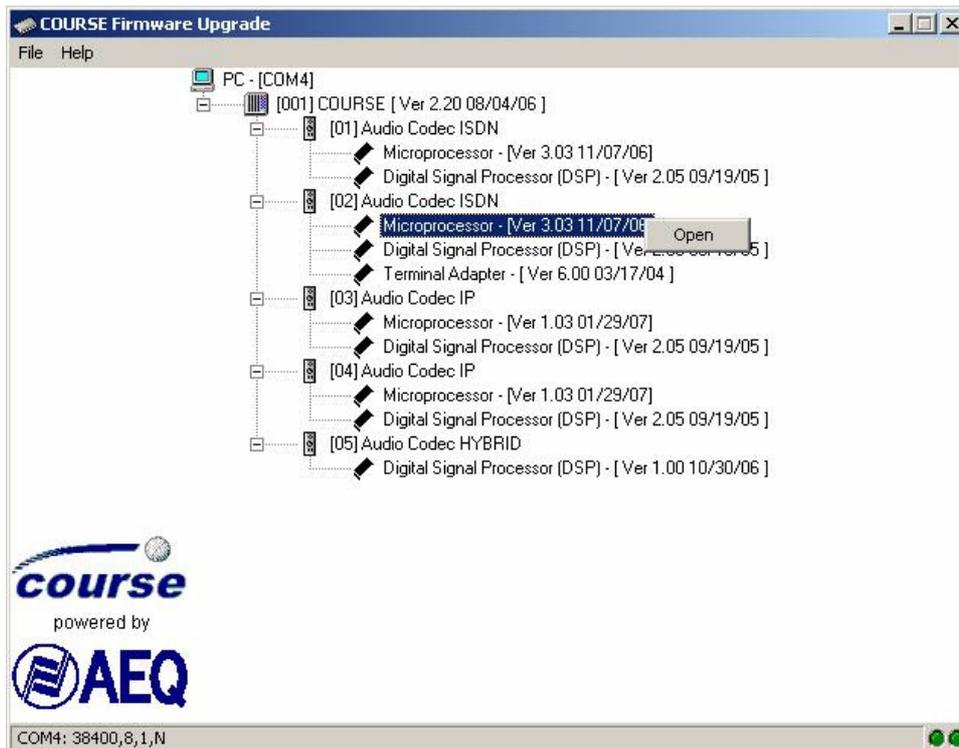
By clicking on the '+' sign of any one of the boards, the structure is expanded, allowing you to see the software modules of the board and its firmware versions:



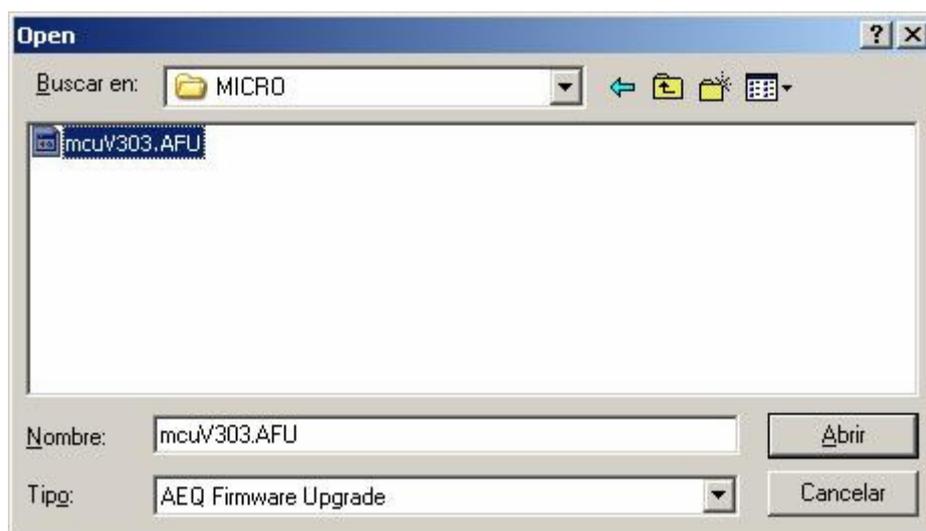
There are four upgradable software modules in this equipment:

- Controller (unique for every equipment, the firmware version is shown close to the symbol of the COURSE).
- Microprocessor (only for ISDN and IP boards).
- Digital Signal Processor - DSP.
- Terminal Adapter - TA (only for ISDN boards, it's an optional module).

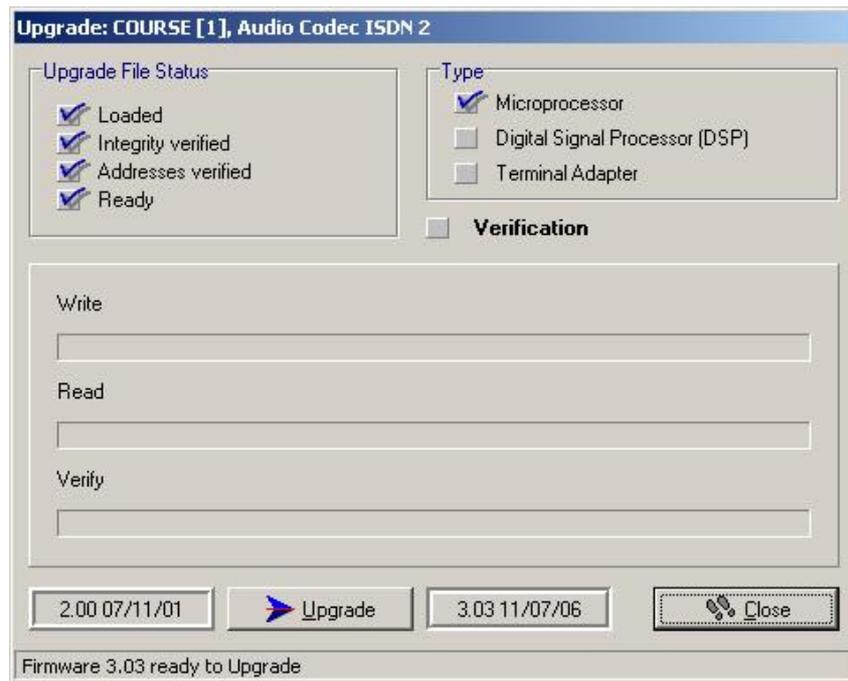
To upgrade a module, you must place the mouse cursor over it and click with the right button. The “Open” option will be displayed:



After clicking on “Open” (left button), a new window will appear where the program asks for the address of the file with the new software version:



Once the file path has been specified, the upgrade window opens up:



In this example, the software has detected version 2.00 of a ISDN board microprocessor and is ready to upgrade it with version 3.03. If this is correct, press **Upgrade**, and if you do not want to upgrade it, press **Close**.

Once writing is complete, the board will restart and a message will be displayed on the screen indicating that the upgrade is complete:



Firmware upgrading order:

TA - DSP - MICRO for ISDN boards.

DSP - MICRO for IP boards.

DSP for **hybrid** boards.

It's very important to upgrade the modules in the indicated order. When all the boards are upgraded, then **controller** board must be upgraded.

If there have been hardware changes (boards have been added or their position has changed), it's necessary to upgrade **twice** the **controller** board. This action is repeated in order to erase the memory of system boards configuration and update it.

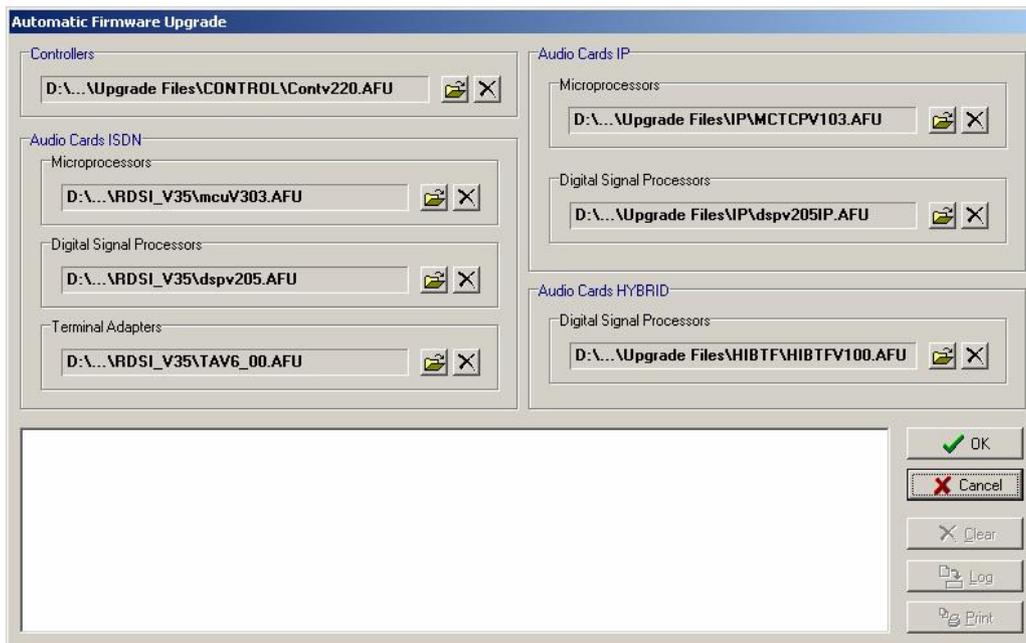
NOTE: It is also recommended that all versions of all the cards installed are the latest ones. Consult AEQ website (<http://www.aeq.es>) or contact Technical Assistance Service (sat@aeq.es) to find out the latest versions for this equipment y the compatibility between versions.

Automatic firmware upgrade.

The process of upgrading a COURSE with several boards takes some time. This time depends on the power of the computer running the upgrade, but in general this time is fairly long. Nevertheless, it is possible to program an automatic upgrade process of the whole system

It is important to ensure that the upgrading process is not interrupted by any reason. This could cause that one board would lose its configuration, making necessary to send this board back to AEQ in order to make a low level programming.

To start the process, open the **File** → **Automatic** option and fill in the boxes with the new firmware files you want to use for every module of every kind of board. Once it is done, press “**OK**” in order to start the upgrade.



6. COURSE TECHNICAL SPECIFICATIONS.

ISDN - V35/X21 and TCP/IP communication boards.

Analog inputs / outputs.

- Transformer balanced, with built-in RF filters, "HARTMANN" type Strip Connectors with 4 Pins.
- Input impedance: > 6 K Ω
- Maximum level: + 22 dBv.
- 24-bit A/D converters.
- Output impedance: < 50 Ω
- Maximum level: + 22 dBm.
- 24-bit D/A converters.

Digital audio interfaces.

- AES/EBU (AES-3), with transformer.
- Sample rates supported: from 16kHz to 48kHz.
- Sample rate converters (SRC): range 1:3 and 3:1, 24 real bits without truncating, independent on inputs and outputs.
- External AES-11 synchronism input.
- Dynamic range in the SRC: 128 dB.
- THD + noise in the SRC @1kHz: -117 dB.
- Dual AES/EBU interface for independent mono inputs (with different sample rates) and configurable to a single dual input.
- Format: DB-9 connectors.

Hybrid board.

Analog inputs / outputs.

- Transformer balanced.
- Input impedance: > 6 K Ω
- Nominal input level: 0 dBu.
- Nominal output level: 0 dBu.

Telephone line interface.

- Input/Output with transformer.
- Impedance: 600 Ω
- Nominal input level: - 4.8 dBu.
- Nominal output level: - 0.7 dBu.

Bandwidth (extender not active):

Line: 300 - 4.000 Hz +/- 1 dB.

Program output: 300 - 4.000 Hz +/- 1 dB.

Distortion + THDN - 1 KHz with nominal input and output levels:

Line output: < 0.12 %.

Program output: < 0.20 %.

Absolute noise: Tx =< - 60 dBu.

Rx =< - 54 dBu.

Frequency extensor: Tx = 250 Hz.

Rx = - 250 Hz.

Cross-talk in any case: Lower than absolute noise.

Suppression of electrical echo:

Line impedance: 600 Ω

Signal source: 1KHz.

Nominal input and output levels.

Program output level: < - 50 dBu.

Digital audio interfaces.

Same as for ISDN - V35/X21 and TCP/IP communication boards.

Communications interfaces.**ISDN:**

- "S" 2B+D Euro ISDN-compatible interface (ETS 300 012, ETS 300 125, ETS 300102), RJ-45 format.
- "U" 2B1Q ANSI-compatible interface (ANSI T1.601-1992, T1.602-1996, T1.607-1998), RJ-11 format.

Point to point (Leased lines): RS-422 physical interface, connectable to V.35 or X.21, DB-25 format.

ISDN Protocols Available.

- Euro ISDN
- National-1 under 5ESS AT&T, Northern Telecom DMS-100, and Siemens EWSD

Other interfaces.

- Auxiliary data: DB-9, RS-232, asynchronous, 8 data bits, no parity, 1 stop bit, configurable speed.
- Remote control: DB-9, multi-point full-duplex RS-422, 38,400.
- Remote control: DB-9, RS-232.

Power.

Auto-range, from 90 to 250 VAC, 50/60Hz, with automatic power factor correction.

Standards.

Electromagnetic Compatibility EN 50081-1, EN 50082-2.
CE Marking.

Dimensions and Weigth.

4 U Heigth, 176mm (7 inch) x 482mm (19 inch) x 450mm (17.7 inch)
21Kgms.

7. A.E.Q. GUARANTEE.

AEQ guarantees that this product has been designed and manufactured under a certified Quality Assurance System and according to the ISO 9001/2002 Standard. AEQ therefore Guarantees 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. In consequence, return 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 Center. This guarantee does not include the freight charges of the product to or from such Authorized Technical Service Center.

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.

8. ANNEXES.

8.1. IP Connections.

- **To find out a computer's name on a LAN network:**

From the desktop, right click on the 'NETWORK NEIGHBORHOOD' icon and choose the "Properties" option.



Another possibility is to run the '**set computername**' command from the command interface:

```
C:\>set computername
COMPUTERNAME=OFTECNICA05

C:\>
```

According to the example, the name of our computer is OFTECNICA05.

- **To find out a computer's IP address:**

Run the '**ipconfig**' command from the command interface:

```
C:\>ipconfig

Windows NT IP Configuration

Ethernet adapter E100B1:

    IP Address . . . . . : 192.50.11.5
    Subnet mask . . . . . : 255.255.0.0
    Default gateway. . . . . : 192.50.1.2
```

According to the example, our computer's IP address is 192.50.11.5.

- **To check the status of the IP connection:**

The '**ping**' command is used from the command interface, indicating the name or IP address of the other PC. If the connection is established, the remote PC will respond and a message like this one will come up on the screen:

```
C:\>ping OFTECNICA09

Pinging oftecnic09 [192.50.11.5] with 32 bytes of data:

Reply from 192.50.11.5: bytes=32 time<10ms TDV=128
```

This message means that four messages have been sent to the remote computer 'OFTECNICA09', also known as '192.50.11.5,' and all four have been answered in a time of less than 10 milliseconds.

NOTE: The response time will depend on many factors, among them the distance between the two computers on the network. Thus, on a connection through the Internet, the response time will be much greater than with a direct connection.

If no IP connection has been established, the message will be:

```
C:\>ping 172.50.11.5

Pinging 172.50.11.5 with 32 bytes of data:

Request timed out.
Request timed out.
Request timed out.
Request timed out.
```

In this case, you should contact your systems administrator so he can check the network.

8.2. USA TA configuration.

Practical notes for the configuration of the ISDN TERMINAL ADAPTER (TA) in the UNITED STATES and OTHER COUNTRIES with NATIONAL 1 ISDN PROTOCOL.

8.2.1. Notes about the ISDN and the peculiarities of the different types of ISDN.

8.2.1.1. Generalities about the ISDN.

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

Basic links allow the simultaneous establishment of two connections or data channels, each with a capacity of 64Kbps.

Primary links allow 23 64Kbps channels to be established in the USA and 31 64Kbps channels in Europe.

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

The COURSE has an ISDN Terminal Adapter for Basic links (BRI), which gives it a communications capacity of two independent 64 Kbps data channels. COURSE allows these channels to be grouped into a single 128 Kbps channel, doubling the bandwidth and the communications cost.

8.2.1.2. Types of connection to Basic ISDN Links.

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

The U interface connects the telephone exchange with the subscriber. It has two wires, the same as in conventional analog telephony, and only allows a point-to-point connection (a single device connected to the interface).

The S/T interface is obtained from the U interface, passing it through a Network Terminator (NT1) located at the subscriber's premises. It has four wires and allows a multipoint connection of up to 7 devices.

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

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

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

8.2.1.3. Protocol.

Although the ISDN is supposed to be a standard, differences can be found between the protocols used by different countries.

In the USA, there are several protocols that are incompatible with each other; fortunately, however, there is one protocol that is fairly standardized (called **National-1**), which is usually available at all exchanges.

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

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

8.2.1.4. Directory Numbers (DN's).

These are the telephone directory numbers assigned to the line.

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

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

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

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

8.2.1.6. Line characteristics for the USA. Model line request.

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

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

LINE REQUEST MODEL

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

Long distance carriers: MCI, AT&T, Sprint

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

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

Please provide:

- Two SPID numbers.
- Two DNs

8.2.2. Connecting the COURSE to the ISDN line.

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

8.2.3. Configuration parameters for the COURSE TA.

8.2.3.1. Terminal Adapter Configuration Menu.

8.2.3.1.1. ISDN U / ISDN S Selection.

ISDN U should be selected if the U interface has been used (RJ-11 jack marked ISDN-ANSI). If this option is configured incorrectly, the effect will be the same as if we did not have an ISDN line connected ("Inactive" status with National-1 protocol).

8.2.3.1.2. TEI MODE Configuration.

The **TEI (Terminal Endpoint Identification)** is an identification number used in the signaling protocol to identify each terminal connected to the same ISDN line.

The "TEI MODE" parameter allows the assignment of the TEI identifier to be configured as DYNAMIC or FIXED mode. The dynamic mode lets the network assign the TEI. In the fixed mode, the terminal forces a TEI = 0.

The choice of one mode or the other is only important in some special types of lines that only allow one of the two possibilities. In general, this parameter will be left in the DYNAMIC mode.

8.2.3.1.3. PROTOCOL Configuration.

This defines the protocol used by the Terminal Adapter. There are two options: the NATIONAL-1, which is the one most used in the USA, and the EUROISDN, which is the standard protocol in Europe.

When the National-1 protocol is configured, the line status will go through several stages before reaching "Ready." First it will show "Inactive," indicating that the line is in the process of activation, then it will show "SPID pending," indicating that the line has been activated and is in the process of sending the SPID's; and finally, it will show "Ready" if the SPID has been sent correctly, or "SPID error" if the exchange did not recognize the SPID.

When the EUROISDN protocol is configured, the line status will go directly to "Ready." If the EUROISDN protocol is configured by mistake on a line with National-1 protocol, the line status will go directly to "Ready" (it will not go through the SPID-sending stages) and the unit will not work properly (incoming calls may work but not outgoing ones, calls may work on one channel but not on the other, etc.).

8.2.3.1.4. NUMBER LINE 1 and NUMBER LINE 2.

These define the Directory Numbers (DN's) for each of the ISDN line's channels. If the National-1 protocol is being used, their configuration is mandatory. If EUROISDN is being used, their configuration is optional (they can be left blank). If a number is programmed for a certain channel, that channel will only respond to incoming calls made to that number.

If these numbers are programmed incorrectly, the unit will allow outgoing calls but will not receive calls.

Note: In the USA, the directory number to be programmed in the terminal should not contain the area code, i.e., only the seven-digit local number should be used.

8.2.3.1.5. SPID LINE 1 and SPID LINE 2 (only for National-1 protocol).

These define the SPID's for each of the channels. The SPID's are associated with each of the channels and are provided by the phone company. Usually, the SPID's are made up of the "**area code + directory number + 0101**," but there may be many other variations, so it is important to get these from the phone company.

If the SPID's are programmed incorrectly, the line status will show "SPID error" and the unit will not allow calls to be made.

8.3. EUROPE TA configuration.

PRACTICAL NOTES FOR THE CONFIGURATION OF THE ISDN TERMINAL ADAPTER (TA) FOR THE EURO ISDN, TYPE S/T INTERFACE.

8.3.1. Notes about the ISDN and the peculiarities of the different types of ISDN.

8.3.1.1. Generalities about the ISDN.

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

Basic links allow the simultaneous establishment of two connections or data channels, each with a capacity of 64Kbps.

Primary links allow 31 64Kbps to be established in Europe and 23 64Kbps channels in USA.

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

The COURSE has an ISDN Terminal Adapter for Basic links (BRI), which gives it a communications capacity of two independent 64 Kbps data channels. COURSE allows these channels to be grouped into a single 128 Kbps channel, doubling the bandwidth and the communications cost.

8.3.1.2. Types of connection to Basic ISDN Links.

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

The U interface connects the telephone exchange with the subscriber. It has two wires, the same as in conventional analog line (telephony), and only allows a point-to-point connection (a single device connected to the interface).

The S/T interface is obtained from the U interface, passing it through a Network Terminator (NT1) located at the subscriber's premises. It has four wires and allows a multipoint connection of up to 7 devices.

In Europe, phone companies put the NT1 directly on the subscriber's premises, and force access through the S/T interface.

In summary, access in Europe is always through the S/T interface. This is made up of two transmit lines, one for each direction of the communication, and each of the lines must have a load of 100 ohms at each end. At the NT1 end, the NT1 itself gives the load. At the other end, the load may be located at some point on the bus cabling (usually on the connector furthest from the NT1) or inside one of the pieces of equipment connected to it. The COURSE has two internal jumpers that allow you to configure whether the termination will be made inside the equipment or not. By default (factory configuration), these jumpers are set, with the termination being made (100 ohms load) in both directions inside the equipment.

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

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

8.3.1.3. Protocol.

Although the ISDN is supposed to be a standard, differences can be found between the protocols used by different countries.

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

8.3.1.4. Directory Numbers (DN's).

These are the telephone directory numbers assigned to the line.

In the case of the EUROISDN protocol, there may be one or several directory numbers, depending on the configuration of the line; these numbers are not associated with either of the two communication channels, unless this association is expressly made by programming them in the Terminal Adapter. In the case of the COURSE, the directory numbers are programmed in the "NUMBER LINE X" Terminal Adapter configuration options. If the numbers are not programmed in the terminal (NUMBER LINE X options blank), any of the directory numbers can be used to call either of the equipment's channels, with the general rule being that the call will enter through channel 1 if it's free, and otherwise will enter through 2. If a number is programmed for a certain channel, that channel will only respond to calls made to that number.

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

8.3.2. Connecting the COURSE to the ISDN line.

If the physical interface used is the S/T (Europe), the line will be connected to the RJ-45 jack (8 pins) marked ISDN-ETSI on the back of the equipment, using an 8-wire modular phone cable (RJ-45 to RJ-45).

8.3.3. Configuration Parameters for the COURSE TA.

8.3.3.1. Terminal Adapter Configuration Menu.

8.3.3.1.1. ISDN U / ISDN S Selection.

Allows the type of physical interface used for the connection to the ISDN to be configured. If the S interface has been used (RJ-45 connector marked ISDN-ETSI), select ISDN S, and if the U interface has been used (RJ-11 connector marked ISDN-ANSI), select ISDN U.

If this option is configured incorrectly, the effect will be the same as if we did not have the ISDN line connected ("Inactive" status with National-1 protocol and "ISDN Link Error" message when trying to make a call with EUROISDN protocol).

8.3.3.1.2. TEI MODE Configuration.

The **TEI (Terminal Endpoint Identification)** is an identification number used in the signaling protocol to identify each terminal connected to the same ISDN line.

The "TEI MODE" parameter allows the assignment of the TEI identifier to be configured as DYNAMIC or FIXED mode. The dynamic mode lets the network assign the TEI. In the fixed mode, the terminal forces a TEI = 0.

The choice of one mode or the other is only important in some special types of lines that only allow one of the two possibilities. In general, this parameter will be left in the DYNAMIC mode.

8.3.3.1.3. PROTOCOL Configuration.

This defines the protocol used by the Terminal Adapter. There are two options: the NATIONAL-1, which is the one most used in the USA, and the EUROISDN, which is the standard protocol in Europe.

When the National-1 protocol is configured, the line status will go through several stages before reaching "Ready." First it will show "Inactive," indicating that the line is in the process of activation, then it will show "SPID pending," indicating that the line has been activated and is in the process of sending the SPID's; and finally, it will show "Ready" if the SPID has been sent correctly, or "SPID error" if the exchange did not recognize the SPID.

When the EUROISDN protocol is configured, the line status will go directly to "Ready."

If the NATIONAL-1 protocol is configured by mistake on a line with EUROISDN protocol, the line status will continue to show "Inactive" or "SPID pending," and calls cannot be made.

8.3.3.1.4. NUMBER LINE 1 and NUMBER LINE 2.

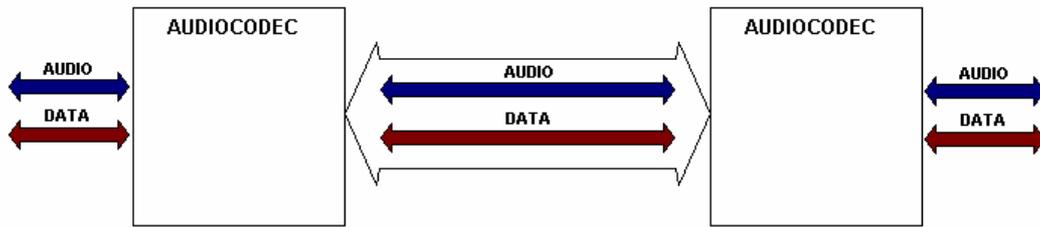
These define the Directory Numbers (DN's) for each of the ISDN line's channels. If the National-1 protocol is being used, their configuration is mandatory. If EURO ISDN is being used, their configuration is optional (they can be left blank). If a number is programmed for a certain channel, that channel will only respond to incoming calls made to that number.

If these numbers are programmed incorrectly, the unit will allow outgoing calls but will not receive calls.

8.3.3.1.5. SPID LINE 1 and SPID LINE 2 (only for National-1 protocol, not applicable for EURO ISDN protocol).

8.4. Data channel operation.

AEQ audio codecs have the capability of inserting a configurable, two-way, transparent end-to-end auxiliary data channel in communications that have been established.

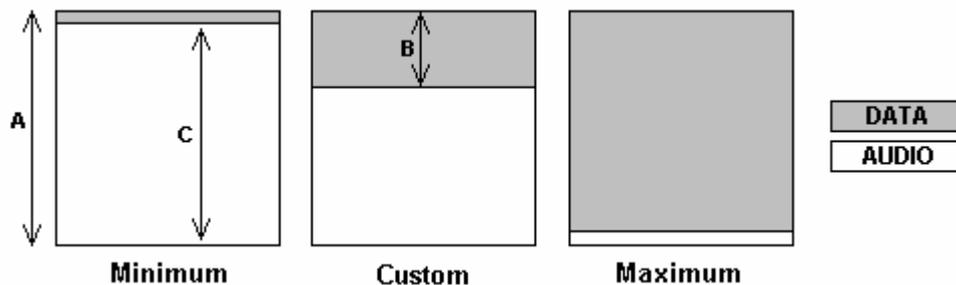


The link used by audio codecs on which the data channel is defined may be ISDN (64 Kbps or 128 Kbps), or using the V35/X21 interface (64 Kbps, 128 Kbps or 256 Kbps in some models) connected to modems or external terminal adapters.

AEQ's first generations of audio codecs (ACD3001, ACD5001, MPAC-02, TLE-02D) already had this feature incorporated, but they only had a limited number of binary speeds that could be assigned to that data channel (300, 1200, 4800, 9600 bauds).

The new generation of audio codecs (EAGLE, COURSE, SWING) improves and expands the use and configuration capabilities of the data channel.

The new AEQ audio codecs use the capacity of the channel available for the transmission of audio circuits, and are also able to model a transparent data channel, as represented in the following diagram:



A: Represents the channel capacity or bandwidth available in the link. It may be 64 kbps, 128 kbps or 256 kbps.

B: Represents the effective bit rate of the data channel or the part of the bandwidth available in the link used for inserting the data channel (configurable by the user).

C: Represents the bit rate available for transmitting audio. The audio codec transmits the audio, compressing it according to the available bit rate.

- If data channels with reduced binary rates are used, there is no perceptible degradation of quality in the audio signal transmitted.

Port configuration.

The external device that transmits and/or receives data through the data channel will be connected to the audio codec using the RS232 ANCILLARY PORTS; a standard RS232 cable like this will be used:

<u>PC</u>	<u>COURSE</u>
Tx	Tx
Rx	Rx
GND	GND
CTS	CTS
RTS	RTS

- Consult ANCILLARY CHANNELS pinout in chapter 3.2.2 of this manual.

Hardware configuration.

There are two operating modes of the data channel.

a) Variable –software controlled- speed: The serial data port of the external equipment must be configured at the maximum speed (115200 bauds). Software will determine the effective speed of the data transmission.

External equipment should be configured in the following way:

- **115200** bits per second. (Note that this speed is fixed and independent of the bit rate of the (A) channel and of the effective transmission bit rate of the data channel (B))
- **8** data bits
- **1** stop bit
- **No** parity
- Flow control: **Hardware**

When a PC is used as a data terminal, the audio codec can be connected directly to a COM serial port. If another type of device that has a configuration other than the one specified is used, it will be necessary to place an RS232 BAUD RATE CONVERTER between the audio codec and the device. This device can be easily found in the market.

b) Fixed speed: Physical connection between the EAGLE and the external device will be done using a fixed baud rate equal to the effective baud-rate of the transmission, that can be 1200, 2400, 4800 or 9600 bauds.

Software configuration.

The software configuration of the AEQ AUDIO CODECS data channel is performed by selecting:

- The effective bit rate.
- Enabling the data channel: Tx or Rx direction, or both.

To select the effective bit rate in the variable speed configuration, there are three possible modes, accessible in the configuration menu: Minimum, Maximum and Custom. The latter is programmed from the control software in real time, and is a value between the minimum and the maximum value defined for each case. For the fixed speed connections, the selection between 1200/2400/4800 and 9600 can be done in the same menu.

Enabling or disabling the data channel is also available in the configuration menu of the audio codecs.

In summary:

- The external equipment will be connected with a fixed baud rate of 115200, 9600, 4800, 2400 or 1200 bps to the ANCILLARY port of the audio codec.
- The audio codec will establish a permanent link at 256 Kbps, 128 Kbps or 64 Kbps with the remote audio codec, selecting an operating (audio encoding) mode that allows the use of the data channel.
- Configuration of the data channel will be done selecting the effective bit rate, and enabling Tx, Rx or both.

AEQ audio codecs:

The **EAGLE** audio codec has TWO independent ancillary ports. This means that both ports can operate simultaneously.

Ancillary data port1 is available in the following modes:

- MPEG128, MPEG256,
- MPEG64, G722/H221 and AEQ3.5 on circuit 1

Ancillary data port2 is available in the following modes:

- MPEG64 on circuit 2

In G711, G722 STAT and AEQ LD-2 audio encoding modes, the auxiliary data channel is not available.

The **COURSE** has a SINGLE ancillary port for each audio codec card, and is available in the following modes:

- MPEG128, MPEG256,
- MPEG64, G722/H221 and AEQ3.5 on circuit 1
- MPEG64, G722/H221 and AEQ3.5 on circuit 2

In G711, G722 STAT and AEQ LD-2 audio encoding modes, the auxiliary data channel is not available.

The **SWING** portable audio codec has a SINGLE ancillary port, and is available in the following modes:

- MPEG128,
- MPEG64, G722/H221 and AEQ3.5

In G711, G722 STAT and AEQ LD-2 audio encoding modes, the auxiliary data channel is not available.

Configuration of the effective bit rate in the different modes:

To select the effective bit rate through the software, there are three possible modes:

Minimum: The data channel has an effective bit rate of 1200 bauds. In this mode, compatibility is maintained with earlier models of audio codecs (ACD5001 and MPAC02) in MPEG 64 Kbps and 128 Kbps modes.

Maximum: The bandwidth for the data channel occupies nearly all of the link's capacity, so the audio transmitted is very low quality or non-existent.

Custom: The effective bit rate in the data channel is programmed by the user through REAL TIME CONTROL software. According to the type of encoding chosen, a menu is displayed with the effective bit rate values that can be selected. The greater the capacity of the data channel, the smaller the channel capacity for the audio circuits, and the lower the audio quality.

In H221-H242 mode, compatibility with the ACD5001, MPAC-02 and TLE-02D is achieved by using CUSTOM programming with a value of 0.

In G722, H221-H242 and AEQ3.5 modes, it is important to select the same effective bit rate at both ends.

Examples:

- A)** When you want to transmit a data channel at **19200** between two PC's through a communication between two **COURSE**, you must:
- Configure the serial port on both PC's at 115200.
 - Program the next higher effective bit rate after 19200 (for example, 20000 on a 64K circuit using MPEG48MONO) with the REAL TIME CONTROL software.
 - Select CUSTOM.
 - Select ENABLE on both codecs (Enable Tx and Rx).
- B)** When you want to communicate two pieces of data equipment that use a **hardware** serial port at **2400** bauds through a communication between two **COURSE**:
- Connect the codecs to the data equipment through the RS232 port.
 - Select 2400.
 - Select ENABLE on both codecs (Enable Tx and Rx).

8.5. Protocols and methods used for audio transmission through IP.

At this moment there are several standards that define the norms for the transmission of multimedia contents (voice, audio, video...) through Internet. The main ones are created by the ITU-T such as H-323, and the IETF that uses Session Initiated Protocol, SIP [RFC-2543]. In both cases the norms defines the set of protocols necessary to conduct this type of communication (transport in real time, signaling, control and establishment of sessions with several participants, maintenance of a certain quality of service QoS, security, additional services, etc.).

It is necessary to consider that these norms are for the creation of telephony networks and multi-conference over IP. In our case we are only seeking to establish a communication of coded audio between two proprietary units, which does not require most of the mechanisms established in the norms.

The most important issues to be solved are:

- Protocol used for the transport of real time audio.
- Protocol for the establishment and maintenance of communications.
- Protocol for the management and maintenance of a certain quality of service QoS.
- Method used for the synchronization.

8.5.1. Transport of audio.

The layer, used for the transport of audio information in real time, is the most important for us, since the quality and capacity of the audio transmission is determined by it. We have selected the RTP protocol, [RFC-3550] which is compatible with H.323 and SIP.

8.5.1.1. RTP protocol.

RTP is based on the use of two protocols: RTP (Real Time Protocol), that is layered over UDP (User Datagram Protocol) and is used for the transmission of audio data, and RTCP (RTP Control Protocol) that is also layered over UDP, but is used for monitoring and control of the transmitted audio.

RTP only adds some characteristics to UDP such as frame counting, time stamp and the type of codification. It inherits the rest of its characteristics. Therefore it is a framed protocol, which is not based on connection; (it does not have to established and maintain point to point connections) it is also unreliable, (UDP frames may not arrive, may arrive duplicated and/or disorganized), but it has the possibility of addressing, (use of ports) and the possibility of broadcast and multicast. The use of UDP also adds the capacity of error control by means of a checksum.

Using UDP means that an equipment can be sending data (in our case, audio), but the receiver may not be receiving the data, either by lack of connection or congestion of the network, etc. In such cases you will hear silence. When the input buffer of the receiver is filled again the receiver will be able to decode that audio. There is no need of any action on the part of the user because the re-connection is automatic.

A special frame has been included into the RTCP protocol, which is sent to the receiver and re-directed to the transmitter to ensure the receiver is decoding the transmitted audio. **'FRAMED' LED** and an alarm in the RTC software allow the user to monitor the framed audio.

The numbering of frames is a fundamental characteristic of RTP that allows the reassembling of the packets of audio and determines which ones are lost. The time stamp is necessary when the system does not send periodic continuous RTP packets (for example, if the periods of silence are not sent): in such case it is necessary to know the temporary position of the audio data.

RTCP is transmitted by the consecutive port used by RTP and adds point to point transmission of statistical information of connection quality and a channel of application data. The main application of RTCP is to be able to make adaptive adjustments of the parameters of audio transport (type of codification, bit rate, redundancy of packets, size of buffers etc.). Based on the quality of the connection; this is especially useful in multi-conference systems where there is the need to adapt to the connection characteristics and capacities of the different participants. RTCP should not consume more of 5% of the bandwidth used by RTP.

For each codification it is necessary to define how to encapsulate the audio data in RTP frames. There are some recommendations for the most standard codifications. The name of the document that describes this is "RTP Payload for Format". In any case, when it is time to define the format the following must be considered:

- In the case of framed codifications of audio, is not advisable to divide the audio frames. This means that RTP frame should be transmitted as one or several complete audio frames.
- The maximum size of the RTP frames must allow it to be encapsulated as a single IP frame.
- To optimize transmission, it is necessary to fill up RTP frames to the maximum possible, but this affects the delay and most importantly the lost of frames. To prevent the degradation of the audio quality, a frame should not contain more than 20 msec of decoded audio.

RTP defines the form in which the frames of audio are transmitted, but it does not establish the mechanisms to solve the problems derived from the use of UDP (frames disorder, jitter in the reception, lost or duplication of frames and lack of synchronism).

8.5.1.2. Solving problems derived from the use of UDP.

The mechanism used to resolve the problems of disorganized frames, jitter in the reception and lost or duplication of frames, is a configurable circular buffer which is organized in blocks, the size of the RTP frames (information of audio) or translated to time units (milliseconds). The time units for the same number of frames will be different for each codification. Thus, a codification MPEG2 generates a frame of 36 msec, a G711 a frame of 16 msec, etc.

The size of this buffer is a fundamental parameter, since the capacity to absorb the jitter introduced by the network and to recover frames that arrived disorganized depends on it. The bigger the buffer, the greater capacity of recovery, but the delay will also be increased.

In the following section you will see that the variation of the position of frames in the buffer will be the parameter used for adjusting the sampling clock (recovery of sample synchronism).

8.5.1.3. Synchronization.

It is inherent that the use of the Internet may cause dropouts on the receive end. The packets of audio arrive asynchronously and with high jitter. Therefore, in the receiver it is necessary to establish some mechanism to correct the problems derived from this lack of synchronism. There are two possible solutions: one is the adjustment of the frequency of the sampling clock of the receiver to match the sampling clock of the transmitter. This can done using a control mechanism. The second option is to use software configured to compensate for the difference by adding or eliminating audio samples.

We have chosen the adjustment of the reference frequency in the receiver since although it necessitated the installation of specific hardware; it is a simpler method with less delay (the only delay is introduced by buffering).

8.5.2. Establishment of communication.

When using the UDP transport protocols, a physical connection does not occur. In order to know if the other end is "listening" RTCP is used, but this only works once the audio transmission has started. In order to verify that there is IP connectivity between both ends, a proprietary protocol has been incorporated which will indicate a "FRAMED" connectivity exists. This must be checked before sending audio. If the receiver is using non-proprietary software, such as Winamp, no indication of connectivity can be established until audio is sent.

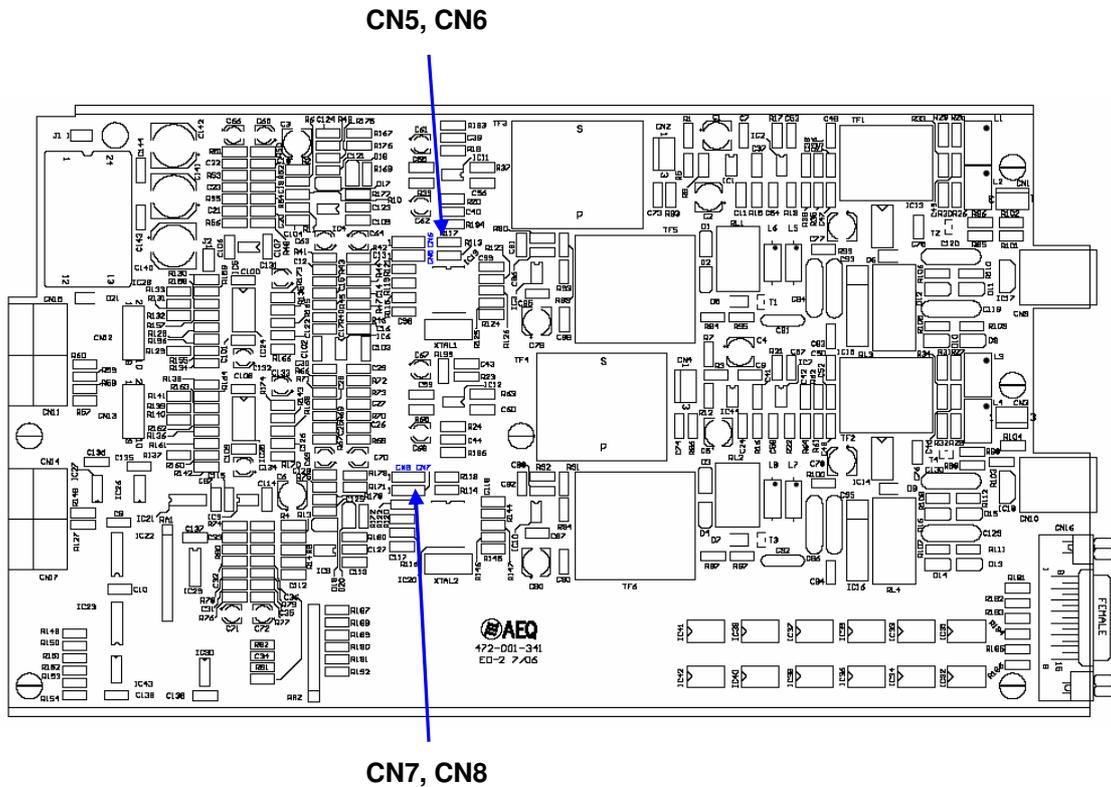
8.5.2.1. Management of the Quality of Service (QoS).

The management of the quality of the service on the Internet is an emerging technology in which two models of implementation exist (integrated services based on reserve of resources and differentiated services based on the differentiation from type of packets). There are also different protocols (RSVP, MPLS, DS, NBC etc...). At the present time there are routers and applications that implement these services on private networks, but its adaptation for use on the Internet is still in development.

At the time of manufacturing, a mechanism of QoS management was not available, but one will be implemented, when it becomes available in the future.

8.6. Incoming call identification in hybrid boards.

The rear card of **COURSE** hybrid boards has jumpers that allow you to configure if the incoming number identification will be compatible with the standards used in Europe or in USA.



Those jumpers are:

- **CN5** and **CN6**: Allow you to configure the type of FSK signal that will be managed in the “Caller ID” of **line 1** of hybrid board.
- **CN7** and **CN8**: Allow you to configure the type of FSK signal that will be managed in the “Caller ID” of **line 2** of hybrid board.

Its configuration is the following:

- **Position 1-2**: compatible with **CCITT V.23** FSK demodulation (Europe).
- **Position 2-3**: compatible with **BELL 202** FSK demodulation (USA).

NOTE: ETSI CCITT V.23 is used in Spain, Eire, France, Germany, Norway, Italy, Turkey and South Africa. BELL 202 or TELCORDIA is used in USA, Canada, Australia, New Zealand, China and Singapore.

8.7. Special features included in the standard control software.

Equipment on screen filtering.

When the different COURSE cards are used from different studios, every user has access to all the codecs from all locations.

It is possible to filter which COURSE boards will appear in the control screen. This mode is useful when many users from different studios have to control different COURSE boards from the same E@sy network.

This option is configured in '**FILTER**' section of the "**Course.ini**" file, in each of the computers where the COURSE Real Time Control application is installed.

In the "**Course.ini**" file will appear this:

```
[FILTER]
INCLUDE=
EXCLUDE=
```

If we would like to see on the screen only the COURSE cards 1, 2 and 3, of a course which node number is 1, it would be:

```
[FILTER]
INCLUDE=1.1,1.2,1.3
EXCLUDE=
```

If we would like to see all the COURSE cards except numbers 9 and 10, it would be:

```
[FILTER]
INCLUDE=
EXCLUDE=1.9,1.10
```

Customizing sounds.

It is possible to change default sounds used in the application modifying the "**Course.ini**" file as showed:

```
[SOUND]
RING=C:\SOUNDS\RING.WAV
SIRENS=C:\SOUNDS\SIRENS.WAV
```

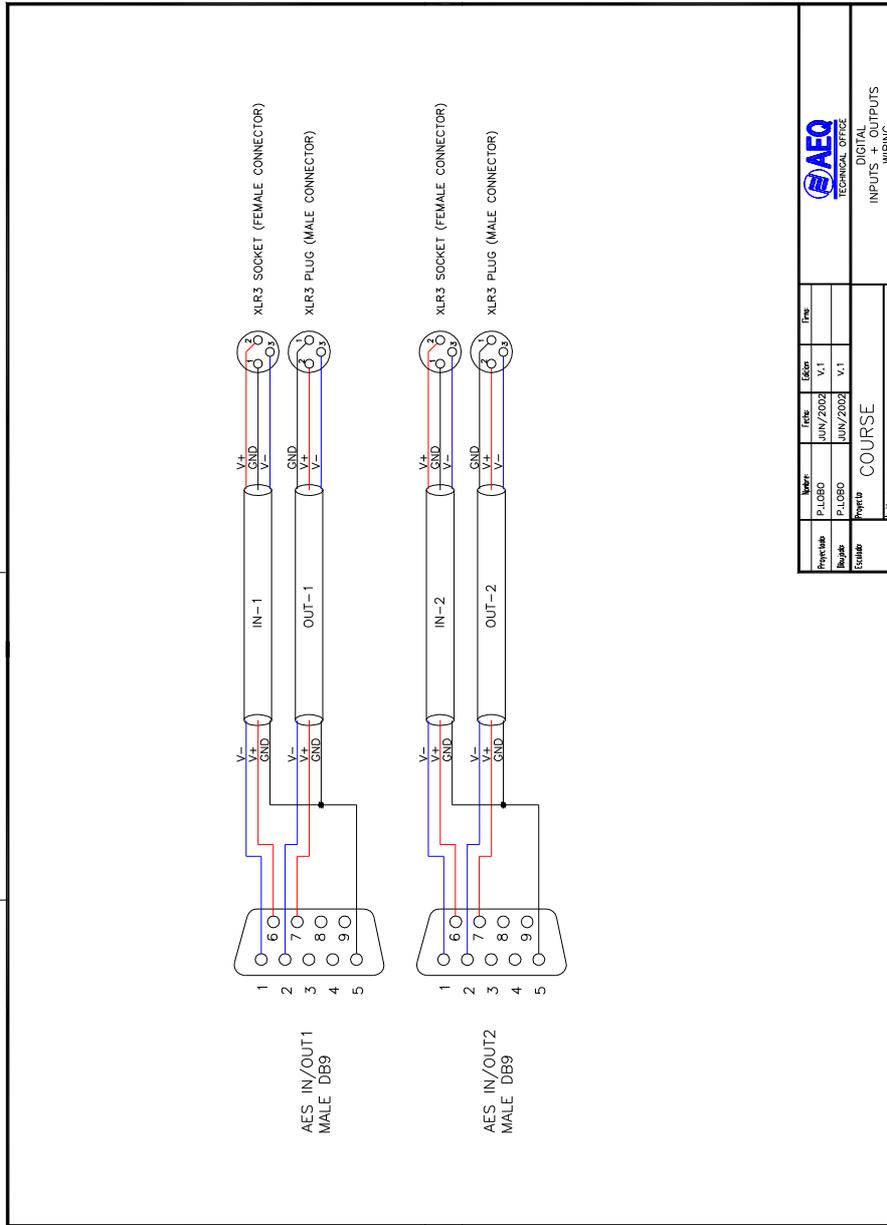
In this example, the wav file 'C:\SOUNDS\RING.WAV' will sound when a call is received and the wav file 'C:\SOUNDS\SIRENS.WAV' will sound when a warning or error happens.

If the file can not be found, the default sound will work.

Audio files should be linear PCM WAV format.

- Sound options should be activated in the application **setup** menu.

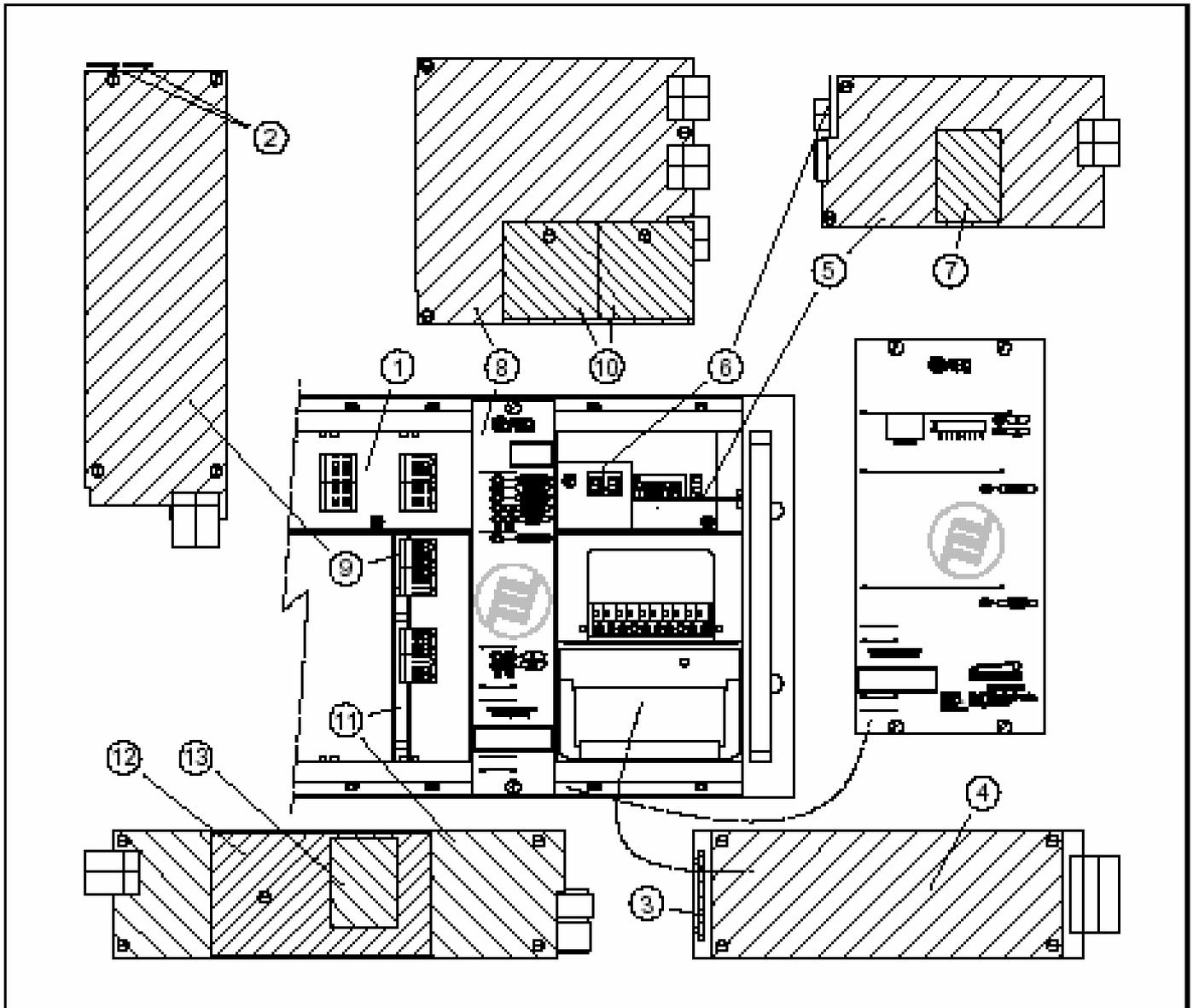
8.8. Standard AES/EBU cable.



NI SE PERMITE LA REPRODUCCION O UTILIZACION DE ESTE DOCUMENTO SIN AUTORIZACION DE APLICACIONES ELECTRONICAS OUSARS

Proyecto	Fecha	Edici3n	Impres
P.LOBD	JUN/2002	V.1	
Revisi3n	JUN/2002	V.1	
Escritura	COURSE		
Proyecto	DIGITAL		
Proyecto	INPUTS + OUTPUTS		
Proyecto	WIRING		

8.9. Parts of the equipment. Boards location.



COURSE-TA ISDN Board and Universal T.A. board (boards **12** and **13** in diagram) are located over the ISDN - V35/X21 communications board (board **11**, connector labeled **CN9**).

COURSE-AES/EBU digital audio boards (**10**) are located in the processor board (**8**). The board that corresponds with **channel 1** is connected in **CN8** and the board that corresponds with **channel 2** is connected in **CN9**.