

OPERATING INSTRUCTIONS AND REFERENCE MANUAL NAGRA ARES-C



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This guarantee covers the repair of confirmed defects or, if necessary, the replacement of the faulty parts, excluding all other indemnities.

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We reserve the right to modify the product, and / or the specifications without notice.

ABOUT THIS MANUAL

This instruction manual is broken down into several sections. The first section is a general explanation of the switches, buttons, connectors of the machine along with the "getting started" section which covers topics such as formatting of the cards and powering of the machine. The second section covers the actual "in the field" operation of the machine for making recordings. The third section is dedicated to the editing of the audio once it has been collected, and the fourth section covers the transmission of the edited audio either by ISDN or telephone, AES or straight analog. The final section of the manual is basically an appendice with some additional information such as the basic theory of different microphones and their selection, the theory of operation of an ALC and an introduction to ISDN.

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1.0 INTRODUCTION

The ARES-C is a 16 bit solid state audio recorder / player using PCMCIA cards as its storage medium. The information is stored using the MPEG layer II, G722, μ -law, A-law compression standards and can be optionally equipped with an on-board ISDN circuit allowing direct connection to the ISDN network. Also equipped with a transformer on the output permitting connection to a standard switched telephone line (PSTN) as well as a built-in editor and weighing less than 3 kg, makes the ARES-C the most versatile reporters tool available.

The front panel, metallic chassis and features were all designed using the experience of previous NAGRA recorders which render the ARES-C easy to operate even in harsh environmental conditions. It is powered by four standard "D" cells available worldwide which will give up to 2½ hours of uninterrupted recording on a single set of four alkaline "D" cells. Using NiCd "D" cells it will operate up to 5 Hours and if the double battery box is used, then standard dry cells will give more than 10 hours of operation.

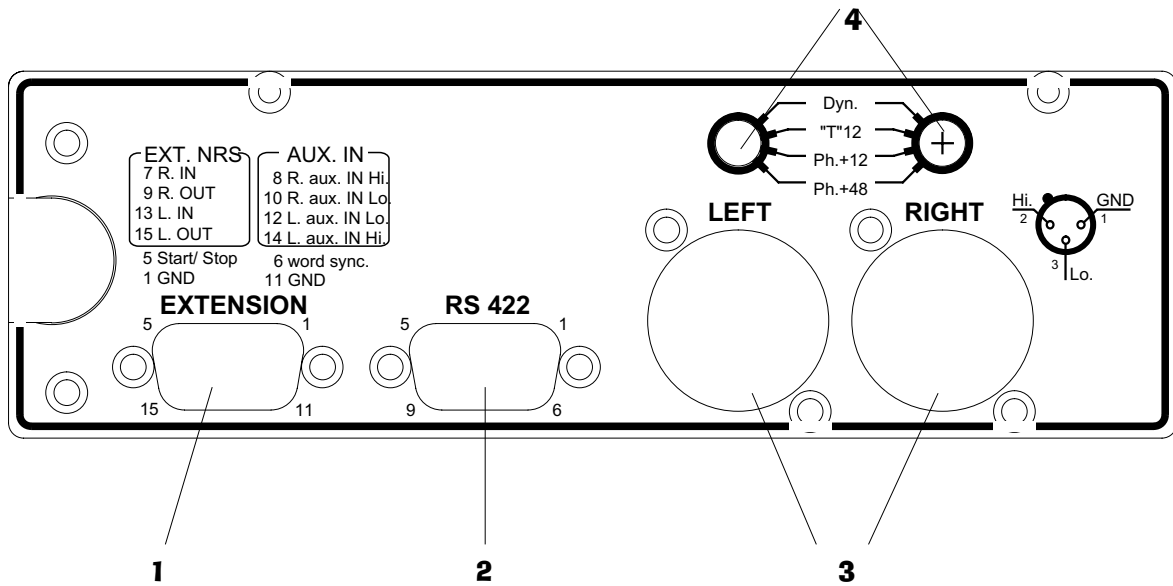
A set of software menus allows simple configuration of the machine for selections such as Mono or Stereo (depending on the compression method selected), ALC programming, machine configuration etc. Equipped with switchable microphone pre-amplifiers and built-in monitoring speaker and headphone output the ARES-C resembles a conventional NAGRA.

The ARES-C automatically generates a directory in the form of an EDL file and has, under its plastic cover, a complete editing keyboard and display so that cut, paste, copy and even hard copy functions are available down to a word length / frame of 24ms. Editing is entirely "virtual" and the edit files are also saved in the card. Different templates will allow you to edit the same recording with several different edit files for different applications (such as a news flash or the evening news or even the late evening discussion).

In addition to the classic analog telephone output, the internal ISDN option gives the ARES-C the opportunity to dial directly to the final destination and to transmit and receive an audio file automatically. ISDN communication data (number called, caller number and tariffs etc) can also displayed on the editor screen (depending on the system employed in the specific country where the ISDN is being used). A full RS 422 communication port gives access to not only diagnostics but also complete remote control possibilities (end '96).

2.0 EXPLANATION OF THE PARTS OF THE MACHINE

2.1 Left side panel



2.1.1 EXTENSION CONNECTOR

This 15 pin "D" type connector serves several purposes. It has a symmetrical transformerless line input (AUX), an external digital word clock input, left and right IN / OUT connections for the use of an external noise reduction system and is wired for start stop option. The connection details printed on the side panel are not entirely accurate the correct pinning of the connector is as follows:

Pin #	Connection
1	Ground
2	Not presently used
3	Not presently used
4	Not presently used
5	Start / Stop - connect this pin to ground to stop
6	External word clock in (square wave 48kHz \pm 75ppm always, +5V)
7	External NRS Right channel IN
8	AUX IN right channel High
9	External NRS right channel OUT
10	AUX IN right channel Low
11	Ground
12	AUX IN left channel Low
13	External NRS left channel OUT
14	AUX IN left channel High
15	External NRS left channel IN

NOTE: If an external noise reduction system is connected to the ARES-C then two switches inside the machine need to be moved. If they are moved then the inputs will not be operational if the external NRS is not connected. These two switches S1 and S2 are on either side of the unused connector J12 on the box mother board behind the modulometer. The normal operating position of these switches is that both are towards the exterior of the machine. That is to say S1 to the left and S2 to the right.

External word clock. No additional settings are needed to use the ext. word clock. The word clock needs to be a 48kHz square wave \pm 75ppm independent on the kind of compression rate used.

2.1.2 RS 422 CONNECTOR

This is a standard 9-pin RS 422 symmetrical serial communication port for connection to the external world. This connector is for the moment only be used by the manufactory production for test purposes.

NOTE: A "lap-top" style PC is not always fitted with an RS 422 port. A converter RS 232 / RS 422 must in this case be fitted to the cable to allow the communication. (ND-PCA # 10540)

2.1.3 MICROPHONE INPUT CONNECTORS

Any type of microphone can be connected to these XLR female input connectors. The sensitivity of the microphone inputs is selected by the switches # 6 on the front panel and the levels can be controlled by the two potentiometers # 5. They are wired according to DIN standard.

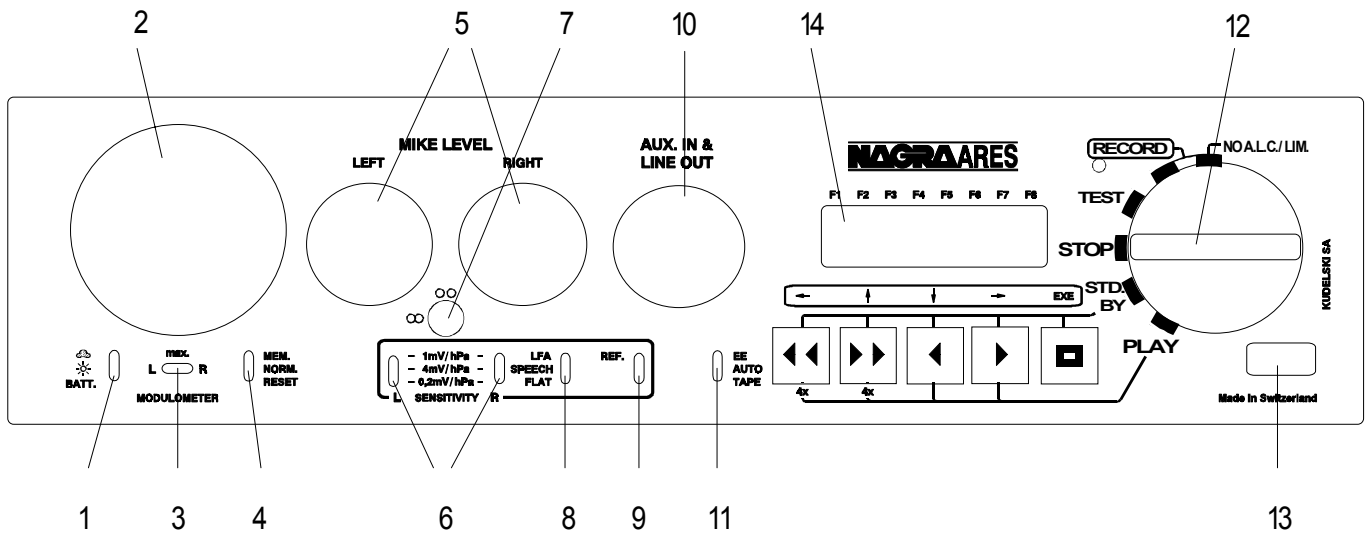
<u>Pin #</u>	<u>Connection</u>
1	Ground
2	Audio signal High
3	Audio signal Low

2.1.4 MICROPHONE POWERING SELECTORS

Each of the microphone inputs can be switched using the selectors #4 on the left side panel of the machine, according to the type of microphone to be used. The possible selections are Dynamic, +12V "T" power or Phantom +12V and +48V. These selectors have a slotted head to avoid accidental changes during operation and must be turned using a screwdriver.

Note: The powering requirements of any particular microphone can be found in their respective documentation.

2.2 FRONT PANEL



2.2.1 LIGHT / BATTERY SWITCH

This three position switch allows the back lights of the modulometer and displays to be turned on/off (cloud / sun positions respectively). In the lowest position the meter will indicate the state of the batteries in the battery box. The indication is given in volts per cell on the meter. The microprocessor of the machine will automatically detect if a single four cell battery compartment or an eight cell battery box is being used, thus the indication is always correct irrespective of the powering in use. All three displays of the machine will light when in the batt. test position. When the cloud position is selected then the display on the editor of the ARES-C will be illuminated for 15 seconds only. If the editor is switched on then it will remain illuminated. This feature allows the operator to have a quick look at the amount of space left in the PCMCIA cards. Pressing to the BATT position will also cause the menus to scroll through their present settings on the front panel display. When the BAT position is selected then the front display will scroll through the presently selected menu settings, the default settings are:

G722	Compression mode selected
ALC OFF	ALC operating mode
AUX OFF	Auxiliary input mode
IN MIX	Input selection
POT OUT	Aux IN / Line pot selection
LEV AUTO	Modulometer Selection
SPK AUTO	Loudspeaker mode selection

More detailed explanations of the possible setting is covered in the MENU section of this manual.

2.2.2 METER

This is a microprocessor controlled moving coil meter which has the ballistics very similar to a modulometer. In normal operation this meter will indicate either input or output levels depending on the menu selection. It can also be used to indicate the state of the batteries in the battery box by means of switch # 1. As the meter is only a single needle meter, it is fitted with two leds, which will indicate the peak values of their corresponding channels when in the stereo mode. The meter scale is calibrated from $-\infty$ to +9 in dB however, if the meter is selected to monitor the input signal and there is an indication above the +9dB point, this means that the A/D converter will be overloaded. The channel being indicated depends on the position of the meter selection switch #3.

2.2.3 METER SELECTION SWITCH

The meter selection switch allows the operator to decide which channel, Left, Right or MAX, will be displayed on the meter. The MAX position is for stereo operation of the machine and the meter will indicate the highest level obtained between the two channels and the leds will indicate which channel this corresponds to. The absence of a second needle, initially thought to have been forgotten, rapidly became accepted as very effective for stereo level adjustment thanks to the two green leds which clearly indicate which of the two channels (Left or Right) is the strongest. This switch is inoperative in mono operation.

2.2.4 MEM / NORM / RESET SWITCH

This is a three position switch. In the NORM position the meter will indicate in the normal manner according to the signal on the input or output (depending on the selection). In the MEM position the highest obtained level (since the last reset) will be indicated. The reset position is a snap-switch position and is used to reset the MEM mode. This switch can be moved freely at any time without affecting the recording. In stereo operation of the machine the operation of this switch is linked to switch #3.

2.2.5 MIKE LEVEL POTENTIOMETERS

These two potentiometers are used to finely control the sensitivity of the microphone inputs. A detailed explanation of the calibration of the scales and their affect on levels and the meter is covered in the appendice to this manual.

2.2.6 SENSITIVITY SELECTORS

These two switches are used to select the desired sensitivity of the microphones connected to the microphone inputs. The possible selections are 1 mV/hPa, 4 mV/hPa and 0.2 mV/hPa. A more detailed explanation is covered under "Sensitivity" in the appendices to this manual. These switches are especially short to avoid accidental modification and need to be operated with a small screw driver or pen.

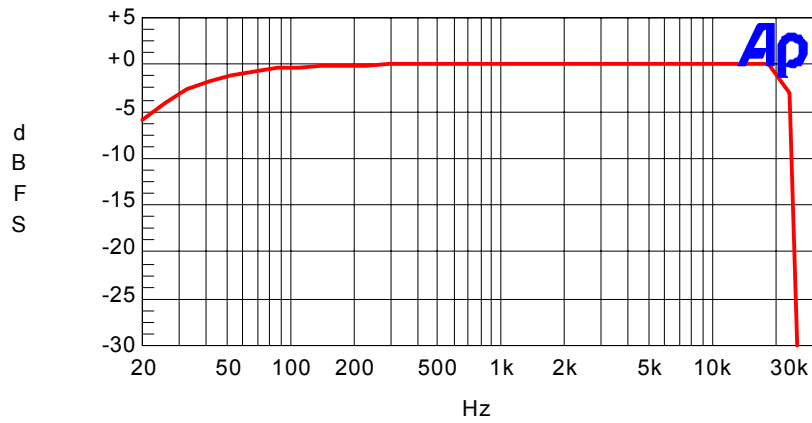
2.2.7 ROTARY LOCK

Used to lock the two mike pots mechanically together. When the button is in the horizontal position "∞" then the two potentiometers are mechanically blocked together irrespective of their individual positions. In the vertical position "∞∞" the potentiometers are totally independent. In order for the button to be moved to the horizontal position it must be depressed slightly.

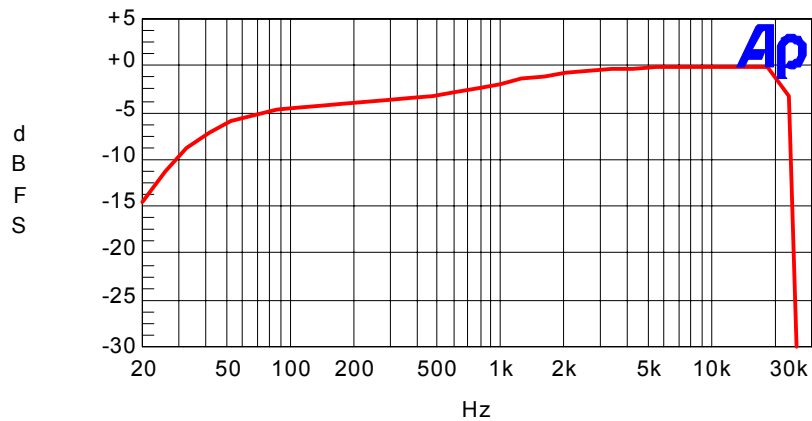
2.2.8 LFA / SPEECH / FLAT

This is the filter selection switch. The filters available are the same as those on other NAGRA models and act on both the microphone and line inputs. The corresponding curves for the filters are shown:

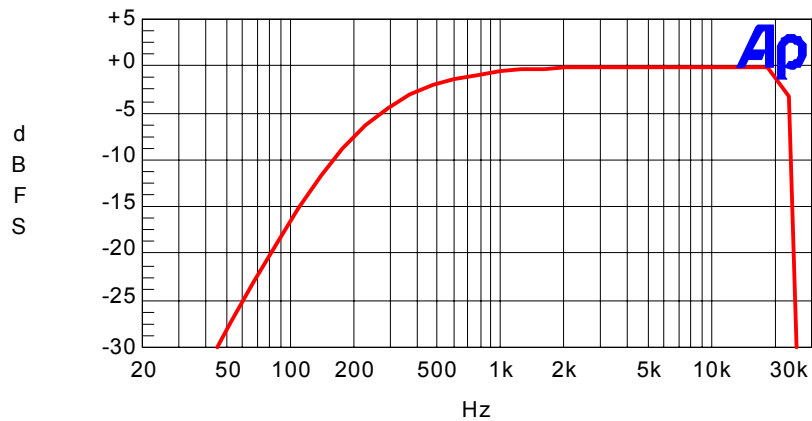
FLAT filter response curve (Measurement at AES bus output)



LFA filter curve (Measurement at AES bus output)



Speech filter curve (Measurement at AES bus output)



2.2.9 REFERENCE SWITCH

Starting from the software version V1.43, the internal reference generator can be activated by pushing the switch upwards. As long as the switch is held in this position, a sine wave of approximately 500 Hz -6 dB, will be present at the outputs, if the ARES-C is in the test position (**the EE / AUTO / TAPE switch must be in the TAPE position**). When the reference switch is held in the upper position during record, the EE / AUTO / TAPE switch may stay in the AUTO position. No indication is given on the modulometer if selected to monitor the INPUT level. While the reference signal is activated, the input signals are muted in the mono mode. In the stereo mode, the reference signal will only be present on the left channel. Only the input signals on the left channel will be muted.

2.2.10 AUX IN AND LINE OUT POTENTIOMETER

This potentiometer has two different functions according to the choice made in the menu mode. If the menu is selected to Line out then this pot will adjust the line output level of both channels simultaneously, as well as the headphone and loudspeaker level. Its position is memorized by the microprocessor of the machine. That is to say, if the pot is set to the 0 dB position, then 0 dB on the meter will give a line output of 1.55V. If the menu is now changed to use this pot to control the AUX input then the initial output setting will be stored in the memory of the machine and will remain at 1.55V. Once the menu is set to the AUX IN mode then this pot serves to adjust the level of the aux line input coming from the 15 pole "D" type EXTENSION connector. Equally if the user changes the use of this pot back to line output, then the previously set AUX IN level will be stored in the memory.

2.2.11 EE / TAPE / AUTO

This switch can be considered as the Tape / Direct switch on a standard NAGRA. In the EE (Electronic / Electronic) position the bandwidth passed by the machine is from 30 Hz to 20 kHz. In the TAPE position the signal that is heard in the headphones is the "off tape" signal meaning that it goes through the A/D, D/A but is not processed (to save battery life the processor works relatively slowly and is not able to compress and decompress at the same time). In the AUTO position, the signal will be in EE mode when the main selector is in TEST position and will be in TAPE mode when the machine is in record or playback. When the internal editor is turned on, then this switch is forced to the TAPE position, and when making a telephone connection then the return of the line will automatically be heard in the headphones if the switch is in the AUTO position.






2.2.12 MAIN FUNCTION SELECTOR

The rotary main function selector is the principle operating switch for the ARES-C. It is a six position rotary selector. Operation of each position is explained below: Each time the main selector is moved from the STOP position the present settings of the menus of the machine will be scrolled through on the front panel display.






- | | |
|------|--|
| STOP | This is the main "OFF" position of the machine. None of the circuits of the machine are powered in this position. When this position is selected the machine will switch off after a few seconds. If the editor was in operation when the main selector is set to the OFF position then the current edit will automatically be saved as the next take number before the machine turns off. In other words it will not switch off if the machine is in the process of writing to one of the PCMCIA cards. |
| TEST | In this position all the circuits are powered allowing the adjustment of levels and signal monitoring. This can be considered as a "stand-by before record" position. If the ALC is ON in the menu mode then the monitored signal will be the signal after passing the ALC circuitry. All menu verification and settings can be made in this position. When the TEST position is selected then the front panel display will scroll through the presently selected settings of the menu tree. |

RECORD There are two different record positions, the first is marked simply RECORD and is the standard position used for recording where the internal ALC circuit will be used providing it has been activated in the menu mode. The second position is marked NO A.L.C./ LIM. which is the position where the internal ALC circuit is deactivated during record. This can be rapidly selected if sudden difficult acoustic conditions occur. (A detailed explanation of the operation of the ALC circuit is covered in appendices to this manual). The ALC will be activated according to the settings programmed. When recording in either position the red led beside the main function selector as well as the write led of the corresponding card in use will be alight. When recording, pressing the grey STOP key will automatically create a new take number without interruption in the recording process. If the card being recorded on becomes full then the ARES-C will automatically continue without interruption on the other card providing there is space. In this event a new take number will be generated on the second card.

EDIT / STDBY This is the position allowing also access to the upper deck functions. (Early machines are marked EDIT and later models are marked STD BY). In this position the grey push-button switches are activated and will act for rewind, fast forward, Skip then stop in both directions and STOP features. When the main selector is put into the EDIT / STD BY position the internal editor of the machine can be switched on and all operations of the editor are made using the keys on the deck plate of the machine. PART III of this manual covers detailed operation of the editor.

-  Rewind at four times nominal speed.
-  Fast Forward at four times nominal speed
-  Skip back by one take and then STOP. The first time this is pressed it will skip to the beginning of the current take.
-  Skip forward by one take and STOP.
-  STOP during rewind or fast forward.

PLAY This is the normal PLAYBACK position. The ARES-C will go into playback mode either from where the machine was after the previous play, or from the beginning of the last recorded take if the machine had previously been in record mode on the card currently selected. Once the play mode has been selected the five grey push-button switches below the display become active (see below).

-  Rewind at four times nominal speed.
-  Forward at four times nominal speed
-  Skip back followed by PLAY by one take each time it is pressed. The first time this is pressed it will skip to the beginning of the current take.
-  Skip forward followed by PLAY by one take each time it is pressed.
-  Toggles between Play and Pause.

2.2.13 SHIFT KEY (front panel)

The shift key must be pressed (and kept pressed) in order to move through the menus on the LCD display on the front panel of the ARES-C. When it is pressed the five grey transport keys operate using their shifted ARROW features. As soon as the SHIFT key is released then it will act as an ESC and the display will return to the main display screen chosen. While in the menu mode the STOP key becomes the EXECUTE function. A full description of the menus is explained later in this manual.

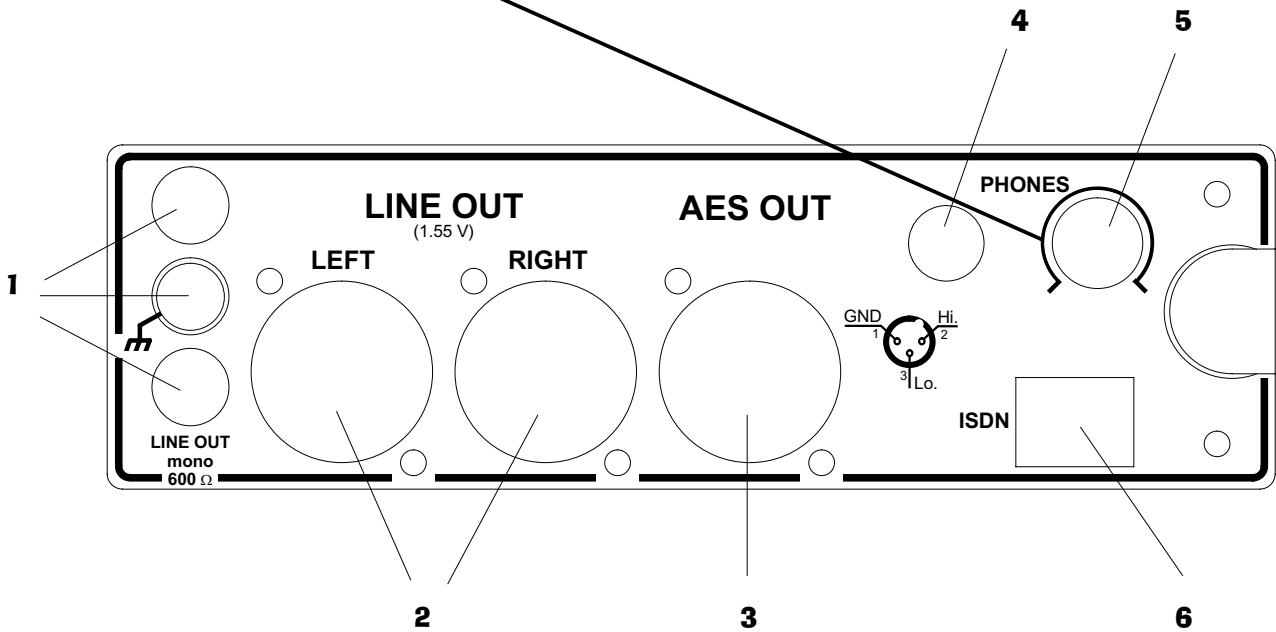
2.2.14 LCD DISPLAY (front panel)

This is a 14 segment 8 digit back lit LCD display, permitting alpha-numeric indication of a large quantity of different information and allowing internal settings of the machine to be made in the MENU mode. In normal operation it will indicate the current take number and time from the beginning of it. It is also used to display the internal STATUS of the machine, ALC status, remaining time available in the card to be recorded etc. The display will be illuminated if the illumination switch # 1 is put in the "cloud" position.

It can be used to display the following:

- Menu Tree
- Take Number and time from start of take
- Remaining Time on selected card
- ALC level

2.3 RIGHT SIDE PANEL



2.3.1 BANANA OUTPUT CONNECTORS

This is the telephone output connection. It is a mono output fitted with a transformer with an output impedance of 600 Ω from 300 Hz to 5 kHz, and is used for connection to a standard switched telephone line. The output level of this connection can be selected in the "TEL LEVEL" position of the menu mode to be either 1.55V or 4.4V. When in operation, the return feed from the telephone can be heard in the headphones or on the internal loudspeaker if selected.

2.3.2 LINE OUTPUT CONNECTORS

These two 3 pole XLR female connectors are the standard analog audio transformerless outputs. The level of which can be controlled by the Line output potentiometer on the front panel (providing it has been previously selected). The nominal output level on these connectors is 1.55V for 0 dB on the meter.

<u>Pin #</u>	<u>Connection</u>
1	Ground
2	Audio signal High
3	Audio signal Low

2.3.3 AES OUTPUT CONNECTOR

The 3 pole male XLR AES output connector is a digital output corresponding to the format of the AES bus used throughout the professional audio industry. The resolution is of 16 bits irrespective of the compression mode being used. This connection allows direct connection to any other digital equipment equipped with an AES interface. The AES output is only available if the compression system chosen is MPEG-1 layer II at a sampling frequency of either 32 or 48 kHz (settable by menu).

2.3.4 HEADPHONE OUTPUT JACK

This is a standard ¼" Stereo Jack connector. The level of the headphone output can be adjusted using the headphone level control. When the ARES-C is connected to a standard telephone line or and ISDN line the return feed of the line is always available in the headphones.

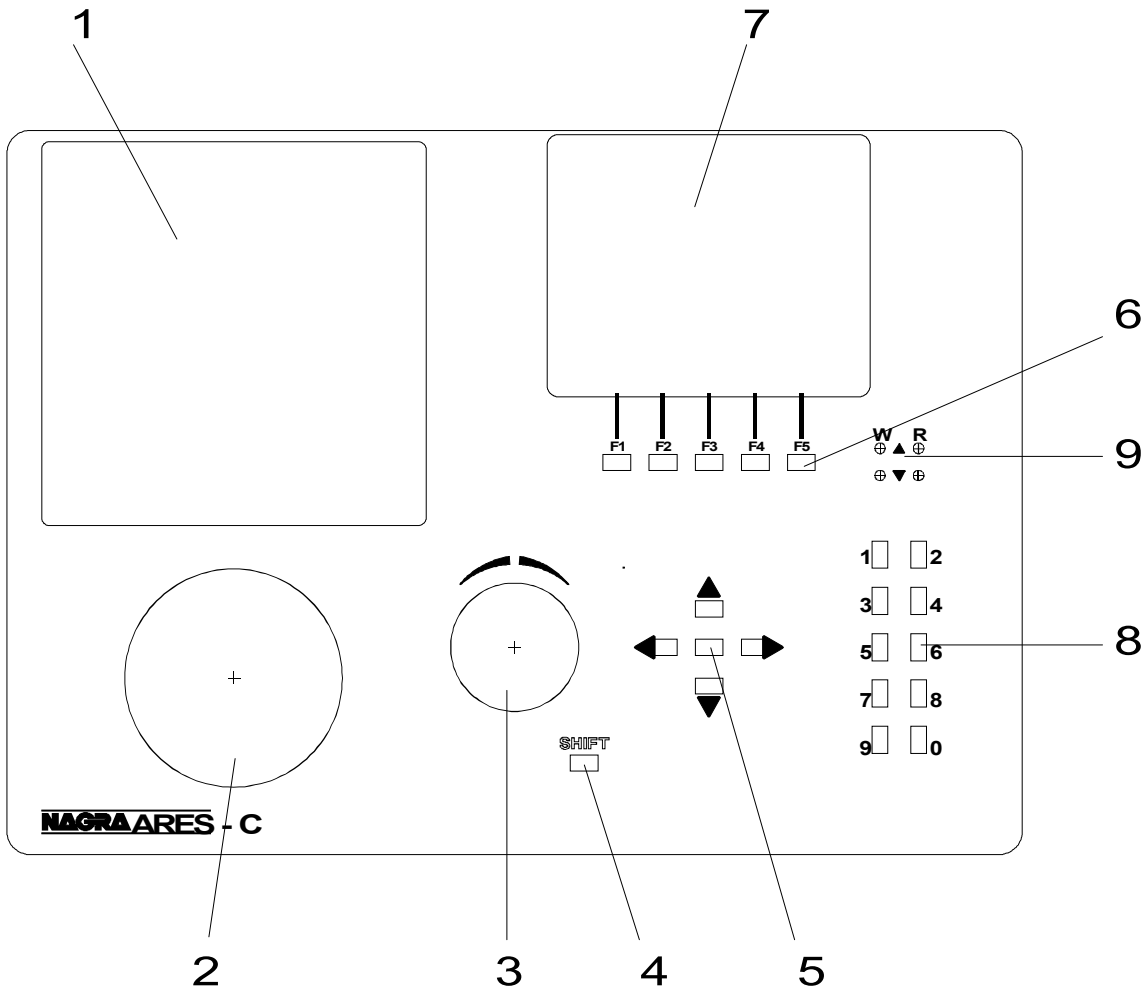
2.3.5 HEADPHONE LEVEL CONTROL

Rotary volume control for the headphones.

2.3.6 ISDN CONNECTOR

If the ARES-C is fitted with the internal ISDN option then this is the connector where the RJ 45 plug is placed to connect the machine to the ISDN network. Operation of the ISDN is covered in detail in CHAPTER IV of this manual as well as in the appendice.

2.4 TOP DECK



2.4.1 DOUBLE PCMCIA CARD SLOT

This is where the removable PCMCIA cards are installed. This slot can hold two cards, the upper position being SLOT A and the lower being SLOT B. The ARES-C will work with cards up to 2 Gbytes which conform to the FLASH type SERIES 2 or SERIES 2+, Strata flash cards up to 192 MB and ATA FLASH cards .

2.4.2 INTERNAL SPEAKER

This small built-in loudspeaker can be used to listen to the recordings and during editing. The volume of the internal loudspeaker is controlled by the Line Out level potentiometer on the front panel of the machine in conjunction with the headphone level pot. The speaker can be switched ON or OFF in the menu mode.

2.4.3 JOG WHEEL

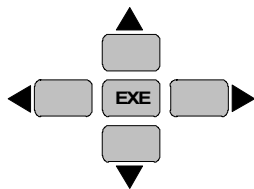
During editing, the jog wheel behaves like a jog wheel of an analogue tape recorder that is to say that turning it clockwise the "tape" will advance at a rate relative to the speed of rotation and similarly in the other direction, allowing the accurate location of edit points etc. The direction of rotation of the Jog Wheel can be selected to either NORMAL or REVERSE in the OTHERSET position of the menu mode. The Jog Wheel can also be used in conjunction with the SHIFT key on the top deck to move through a section of tape at high speed when the internal editor is turned on.

2.4.4 SHIFT KEY (deck plate)

The SHIFT key has several different operations depending on the mode of the editor.

- SHIFT+ ▲ Move directly to the top of the directory listing, when the directory of the card is displayed.
- SHIFT + ▼ Move directly to the bottom of the directory listing, when the directory of the card is displayed.
- SHIFT + F1 Will momentarily display CPY meaning copy, the display will ask the question Copy X to Card Y ? This is used to copy a selected take from one card to the other.
- SHIFT + Jog If the shift key is pressed while the JOG wheel is being turned then the editor will play through the displayed portion of the "tape" at high speed in both forward and reverse directions.
- SHIFT + 0 To increase the deck display contrast, press the shift key with the numeric key 9.
- SHIFT + 9 To decrease the deck display contrast, press the shift key with the numeric key 0.

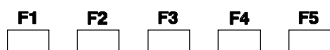
2.4.5 ARROW KEYS (deck plate)



This is a combination of 5 different keys. The central square key is the EXECUTE key and the other four are direction buttons. The four arrow-keys are "auto repeat" meaning that if they are kept pressed then they will scroll in the corresponding direction. The specific functions of each of these buttons are as follows:

- ▲ Move the cursor upwards through the tape directory listing or to move from the edited section of tape to the original source section of the "tape" while editing.
Move the cursor to card "A" if the editor is OFF.
Reduce the attenuation of the mixed playback during transmission.
Pressing this key while the SHIFT key is held down will move the cursor immediately to the top of the directory listing.
- ▼ Move the cursor downwards through the tape directory listing or to move from the original source section of the "tape" to the edited section while editing.
Move the cursor to card "B" if the editor is OFF.
Switch On the mix. Mode and increase the attenuation of the mixed playback during transmission.
Pressing this key while the SHIFT key is held down will move the cursor immediately to the bottom of the directory listing.
If the Editor is OFF, the up and down arrow keys also permit to jump from one card to the other.
During an ISDN transmission, the up and down arrow keys select the mixing transmission modes as well as the attenuated level of the mixed playback signal.
- ▶ Will scroll the display horizontally to the right in the directory listing and will also move the cursor along the "tape" to the next edit point to the right while editing.
- ◀ Will scroll the display horizontally to the left in the directory listing and is also used to move the cursor along the "tape" to the previous edit point to the left while editing.

2.4.6 FUNCTION KEYS

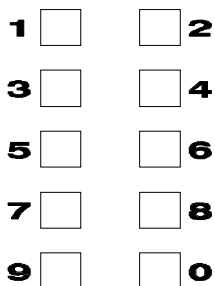


The five function keys located under the display are the principle operating keys for the built-in audio editor. These keys have different functions depending which screen of the editor is being displayed. The operation of each key is indicated on the bottom of the display. (see EDITING for full explanation of each function) A list of the abbreviations for the commands under the function keys can be found at the end of chapter III of this manual.

2.4.7 LCD DISPLAY (deck plate)

The LCD display is a graphic 128 x 64 dot back lit display used for displaying the directory information as well as all the functions of the internal editor.

2.4.8 NUMERICAL KEYS

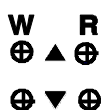


Generally the numerical keys are used for the introduction of user information such as telephone numbers when using the ARES-C with the ISDN option. They may also be used in the directory mode to move the cursor directly to a given take number. All take numbers must be entered in 3 digits (ie take 3 must be entered as 003).

Note: Early machines had the "YES", "NO", "END" and "UNDO" texts next to the buttons. On later machines only the numbers are printed.

In the TRANSMISSION mode, the numerical keys will generate DTMF tone. Once the "TEL." menu or the "LINE" menu is selected, DTMF is activated for as long as the "DIR" menu is not selected. In the ISDN menu, the DTMF tone is activated at the moment that the screen shows "ON LINE"

2.4.9 READ AND WRITE LEDS



These four leds correspond to the Writing / Reading of the two PCMCIA cards. The upper pair (one red and one green) correspond to the upper PCMCIA card "A" and the lower pair correspond to the lower PCMCIA card "B". The RED leds indicate that information is being written to the card and the GREEN leds indicate that the card is being read.

3.0 PREPARATION OF THE MACHINE

3.1 FORMATTING A PCMCIA CARD

In order to switch on the internal editor of the ARES-C set the main function selector to the EDIT / STD BY position and then press the "ON" key F4. If F4 is pressed then the display will now show the software version installed in the machine and four of the five function keys will have a mode written above them.

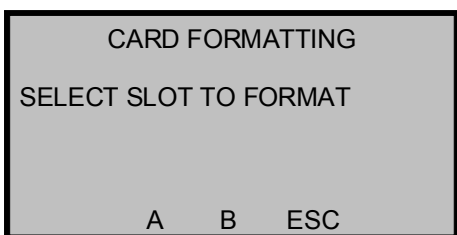


When this screen is displayed, there are five possible choices that can be made by the user "DIR" (directory) by pressing F1, "TRM" (transmit) by pressing F2, "SET" by pressing F3 for internal settings, "OFF" to turn the editor off by pressing F4 or "MIS" (miscellaneous) by pressing F5. DIR will display the directory on the selected card and MIS gives access to various other features. Pressing F5 (MIS) the display will change to the following:



From this screen there are five choices. FOR (FORMAT) "F1" is used to format or reformat a PCMCIA card, CHK (F2) used to recover a corrupted card, RTC (F3) gives access to the internal Real Time Clock and ESC (F4) is Escape as on a PC and will return to the previous display, VER (F5) will show the software versions installed in the machine.

Press F1 (FOR) and the display will indicate the following:



Now select the desired slot corresponding to the card that is to be formatted and the display will immediately indicate "FORMATTING SLOT ___" followed by the % which will increase up to 100% as the formatting progresses. Once the formatting is complete the display will show "FORMAT COMPLETE". At this point another card may be inserted in order to be formatted, alternatively, press ESC to return to the previous menu. The editor must be turned OFF before a recording can be made.

The complete formatting procedure for a 20 Mb PCMCIA card takes approximately 2 minutes and 50 seconds.

NOTE: DO NOT REMOVE THE CARD while it is being formatted or during erase mode. If the batteries die during a formatting then replace the batteries and totally reformat the card. If the machine is switched OFF while formatting is taking place then the machine will finish the format process and will then turn off at the end.

4.0 POWERING OF THE MACHINE (-Ve is ground)

The ARES-C was designed entirely with the "in the field" reporter in mind. Hence the powering is of great importance. The three different ways of powering the machine are standard "D" cells - available almost anywhere in the world, secondly by rechargeable accumulators (same dimensions as "D" cells) or an external +5v to +12V DC supply. The dry cells can be installed in either of the removable battery boxes (NA-BB4 for four cells or NA-BB8 for eight cells). The rechargeable cells can be installed in the NA-DCDC battery box, which also houses the charger circuitry. A rechargeable cell will deliver all its energy even if the current drawn is high. On the other hand, if a dry cell is too heavily loaded, it will only deliver part of its energy. To avoid this situation we recommend using the 8 cell battery box for those who regularly use dry cells, which will give approximately 10 hours of operating time as opposed to about 3 hours with the standard 4 cell box.

To install the batteries remove the battery box on the rear of the machine by lifting the plastic battery box clips on each side of the rear of the machine and remove the battery box. Open the upper lid of the battery box by squeezing the closing mechanism.

Indication of the state of the batteries installed can be seen at any time by selecting BATT on the meter on the front panel of the machine. By means of the BATT menu (see chapter II) it is possible to see how long the installed batteries have been operating. It should also be remembered that the Volts / cell indication on the display of the meter is of little interest when NiCd batteries are being used as the nature of discharge of such cells is very rapid as they become discharged. The microprocessor of the machine will automatically detect if a single four cell battery compartment or an eight cell battery box is being used, thus the indication is always correct irrespective of the powering in use.

When installing batteries into the battery case, be sure that they are installed with the correct polarity according to the sticker inside the battery case (+ve terminals towards the right-hand side of the machine).

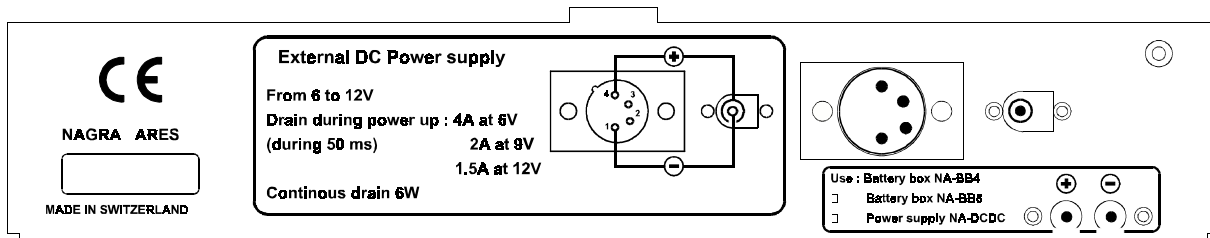
If the external voltage becomes too low then a message "LOW BATT" will be displayed on the front panel of the machine followed by a beep and it will eventually automatically turn off.

NA-DCDC (NiCd battery box / charger)

This accessory can be used with either 4 individual "D" size NiCd cells or alternatively the NA-ACC (#98253) accumulator stick. If the NA-DCDC is connected to the external transformer, but the ARES-C is either OFF or totally disconnected then the NiCd cells inside the NA-DCDC will be recharged. The charging time for four cells (or 1 stick) is approximately 3 Hours (Charge current 1.5A). Charging is automatically stopped when the internal temperature of the cells increases by 10°C with respect to the ambient temperature.

If the ARES-C is switched on while the batteries are being charged, then the power will be supplied by the cells and not by the external supply. When the cells become flat the charger will switch on but in this case will only charge the cells slowly as most of the power being supplied to them will be used by the machine. If the machine is then turned OFF, normal charging will resume.

The green LED on the end of the case indicates that the external DC supply is present. When both the green led and the red led are alight the internal cells are being recharged.

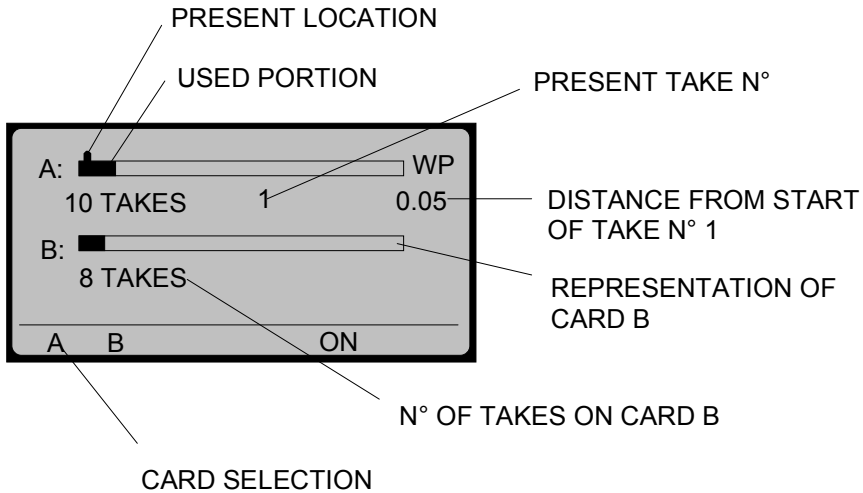


Two other DC power connectors are added to the ARES-C. See above figure for polarity, voltage and current specifications.

5.0 WRITE PROTECTION OF THE PCMCIA CARD

A PCMCIA card has a similar protection as on a floppy disk against accidental recording, in the form of a miniature switch located in the rear edge of the card. This miniature switch can only be moved with a pointed object. For normal operation the switch should be towards the left side (when looking at the card installed in the machine). The position of the Write Protect switch is read as soon as the machine is switched on.

A write protected card is indicated on the display by the letters WP as shown below:



NOTE: If the user tries to format a card when the switch is set to the write protect position then the display will indicate **FORMAT ERROR**.

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QUICK START

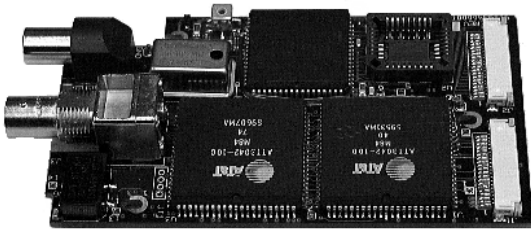
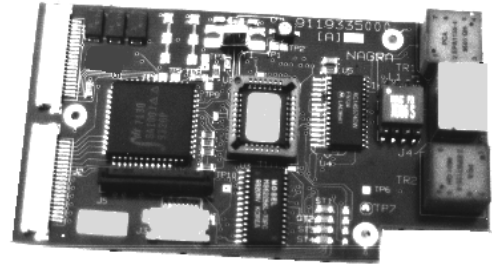
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1.0 DELIVERED WITH.

One ARES-C with a battery box for four "D" type cells
One connector Jack ¼" right-angle for headphones
One user manual
One carrying strap

2.0 OPTIONS NOT INCLUDED WITH THE ARES-C.

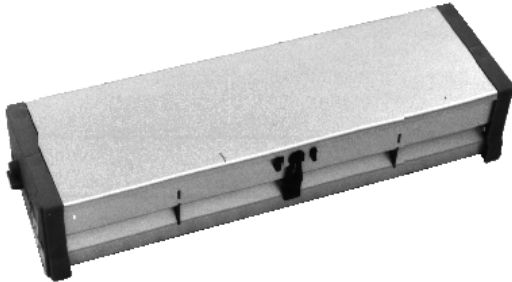
-7019335000 ISDN option



-7019340000 Time code option

-2097194000 Flash card PCMCIA 64MB
-2097197000 Flash card PCMCIA 192MB

-2097195000 Flash card PCMCIA 80MB



-7019110000 Battery box for 8 "D" cells

-7019120000 Battery box for 4 "D" cells (additional)



-7019130000 NiCd charger pack 4 cells
-7019135000 NiCd charger pack 8 cells



-7019145000 AC power supply for NiCd charger pack

-2098253000 4Ah rechargeable NiCd stick



-2099190000 Soft carrying case for ARES-C (if equipped with a 4 "D" cell battery box)

-2099195000 Soft carrying case for ARES-C (if equipped with a 8 "D" cell battery box)



3.0 FIRST NEEDS.

3.1 IF NO ADDITIONAL POWER OPTIONS ARE AVAILABLE.

If "YES" jump to paragraph 3.2

- ◆ Remove the battery box (type 4 "D" cells) from the ARES-C by lifting upwards the two grey side levers at the back left and right side.
- ◆ Open the battery box by pushing together the two finger shape plastic clamps and remove the top cover.
- ◆ Install 4 dry cells type "D" or 4 fully charged NiCd type "D" cells taking care of the polarity indicated and refit the cover again.
- ◆ Remember that the battery box for 4 "D" cells has an optimal application for using NiCd cells and not for dry cells. Anyhow, dry cells can be used but will not be completely discharged.
- ◆ Reinstall the battery box on the backside of the ARES-C and lock the two grey levers again.

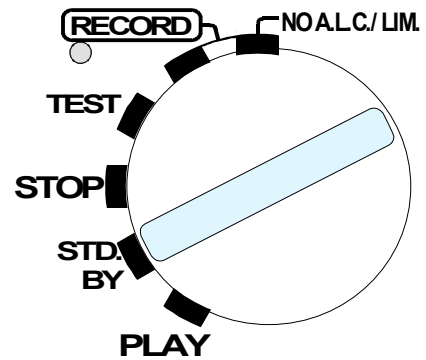
Jump to paragraph 4.0

3.2 IF THE NiCd POWER PACK AND THE AC SUPPLY ARE AVAILABLE.

- ◆ Remove the battery box (4 type “D” cells) from the ARES-C by lifting the two grey side levers at the back left and right side upwards.
- ◆ Install the NiCd charger pack with or without the NiCd cells and lock the two grey levers again.
- ◆ Connect the 2 pin “LEMO” connector from the AC power supply to the NiCd charger pack. Connect the AC power supply to the AC outlet. Once this is done, a green led must light on the NiCd power pack, indicating that a correct DC voltage is received from the AC power supply. If NiCd batteries or a NiCd stick were previously installed inside the charger pack, the red led will light up as well indicating that the automatic battery charge has started.

4.0 SWITCHING ON THE FIRST TIME.

- ◆ Verify that the MAIN selector knob on the front right side of the ARES-C is set to “STOP” (horizontal position).
- ◆ The ARES-C will not switch on if new batteries have been installed and the main selector was not in the “STOP” position. Set to “STOP” after changing the batteries, then wait a few seconds before switching “ON” again.
- ◆ This is also valid if a NiCd battery pack was used in place of the standard battery box.
- ◆ After a few seconds, set the “MAIN” selector to “TEST”. The four LED’s on the deck of the ARES-C will light up for a few seconds. At the same time, the front display will scroll through the main settings of the ARES-C.
- ◆ If no flash card PCMCIA was inserted in the ARES-C, the front display will show a kind of random 8 digit number after scrolling.
- ◆ By pushing the “BATT” switch, located in the bottom left corner of the front panel, the modulometer will indicate the voltage level per cell. Pay attention that if this voltage level drops below 1V/cell, the ARES-C will automatically switch off, but will automatically save any running application such as record or editing.



5.0 SETTING TIME AND DATE.

- ◆ Set the “MAIN” selector on the front panel to the “STD.BY” position.
- ◆ If the contrast on the deck display is not correct, follow the next steps:

SHIFT + 0	Increases the contrast step by step.
SHIFT + 9	Decreases the contrast step by step.

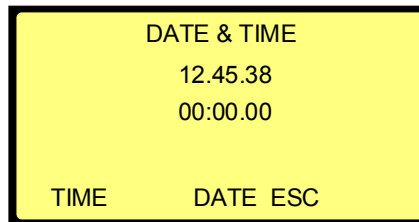
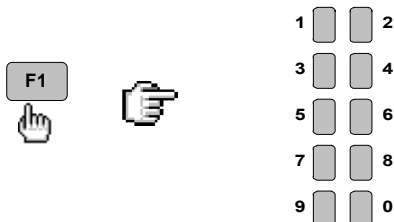
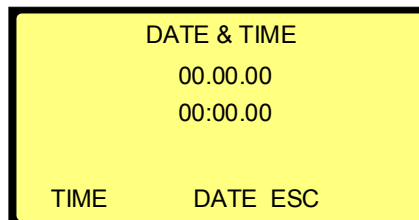
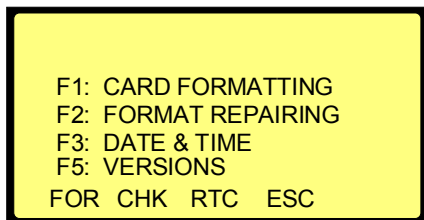
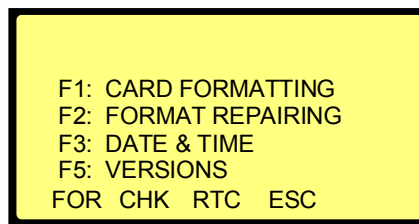
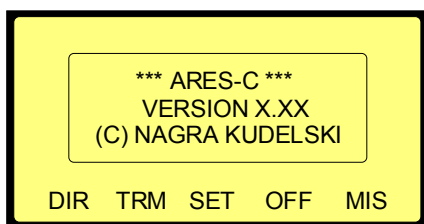
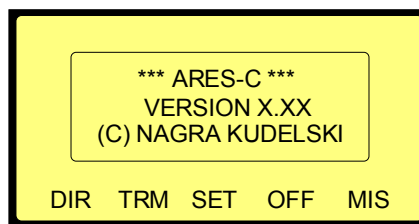
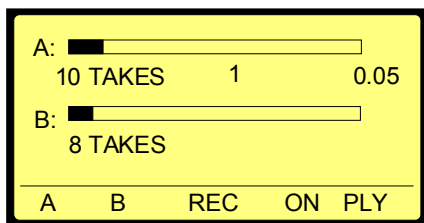
The final result once the “SHIFT” button is released, is stored in the E² prom memory of the NAGRA ARES-C.


A:			
	10 TAKES	1	0.05
B:			
	8 TAKES		
A	B	ON	PLY


OR

DATE & TIME		
	00.00.00	
	00:00.00	
TIME	DATE	ESC

- ◆ Two different displays can be on the screen. The left one is the normal one. If the right display is on the screen, it indicates that MEMORY LOST has occurred due to missing AC power for several hours. To set the DATE & TIME, follow the next steps:

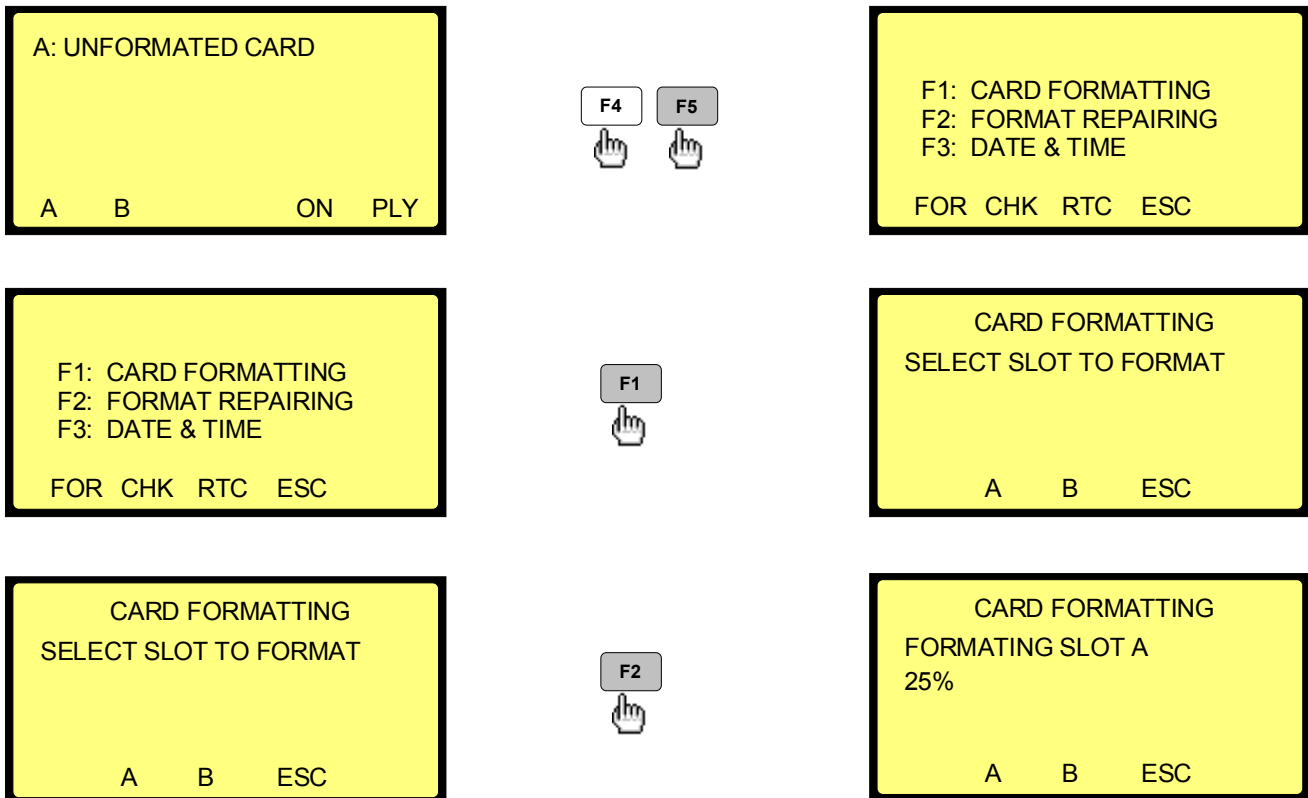


Repeat with  to set the date.

- ◆ To escape from this menu, just press the  button, “ESC” several times until “OFF” appears on the display. Press F4 “OFF” once again, to close the editor.

6.0 INSTALLING THE FLASH CARD AND FORMATTING FOR THE FIRST TIME.

- ◆ Insert the flash card in the upper slot (called "A").
Check that the small switch on the end of the flash card is not in the position "WRITE PROTECT". If it is, then "WP" will be displayed.
- ◆ Set the main selector on the front of the ARES-C to the "STD.BY" position.
- ◆ The display on the deck of the ARES-C will indicate "A: UNFORMATTED CARD".



- ◆ The formatting starts automatically on card A. The display shows a percentage which indicates the status of formatting. Once 100% is obtained, the card is fully formatted. The display will show "FORMAT COMPLETE"
- ◆ Push F4 "ESC" several times until "OFF" appears on the display.
- ◆ Push F4 "OFF" again and the display will show an empty bar indicating "0 TAKE 000 0.00" which means that the card is ready for recording.

7.0 MAKING THE FIRST RECORDING IN MONO USING A MICROPHONE.

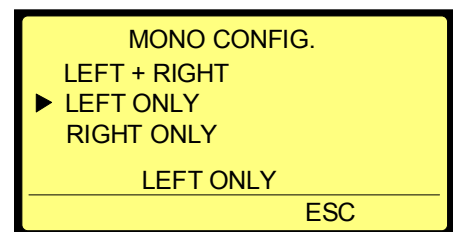
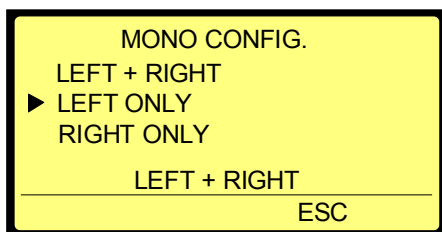
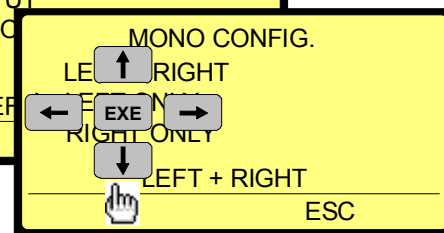
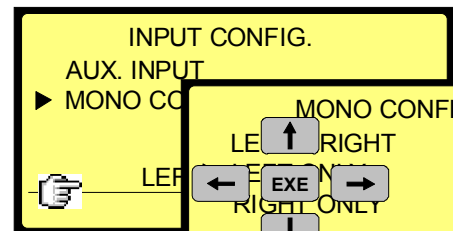
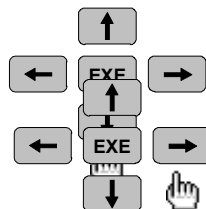
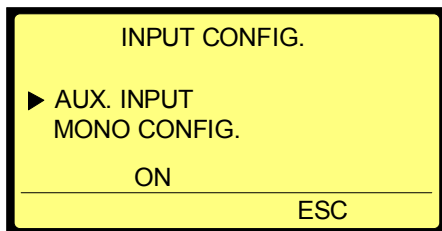
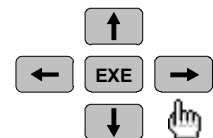
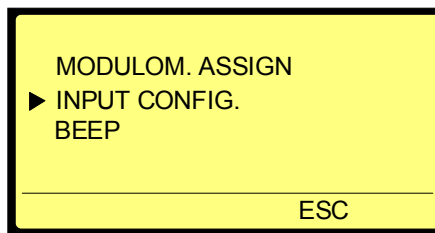
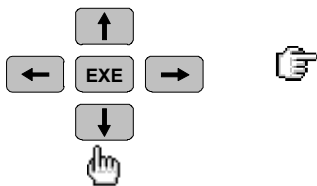
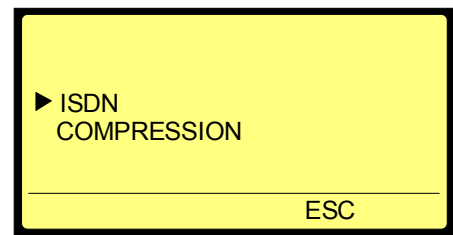
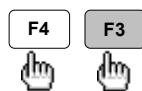
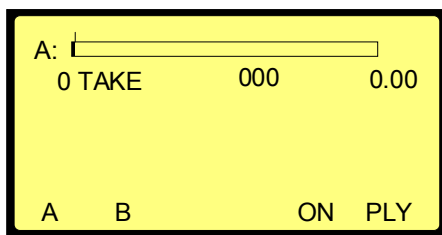
7.1 Microphone connection configuration.

- ◆ Connect a microphone to the left MIC. Input of the ARES-C. Depending on the type of microphone used, select with the switch above the MIC. connector the corresponding power supply (“dyn.”, “T12”, “Ph+12” or “Ph+48”).

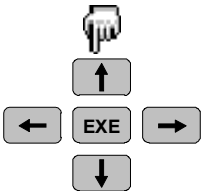
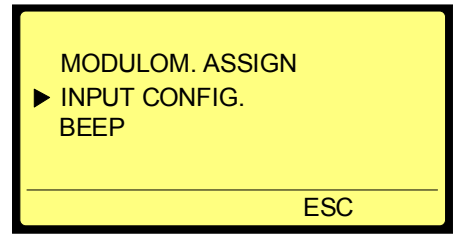
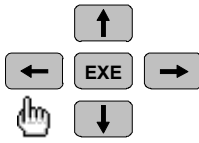
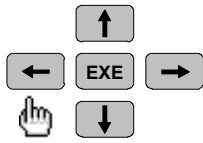
7.2 Microphone input configuration for the left channel.

- ◆ It is possible to record with or without Automatic Level Control. In this example, the recording will be made without ALC and using the G.722 compression mode.

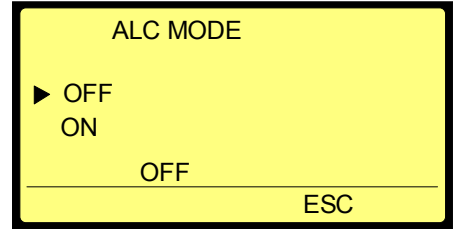
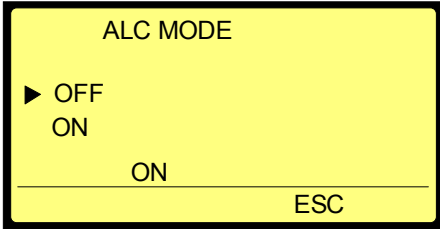
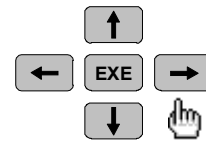
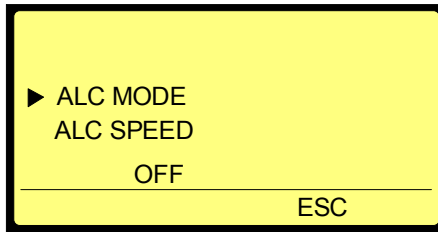
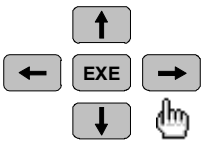
- ◆ IN LEFT selection.



◆ **ALC OFF selection.** The same kind of research through the menu needs to be performed until “ALC” appears:



UNTIL

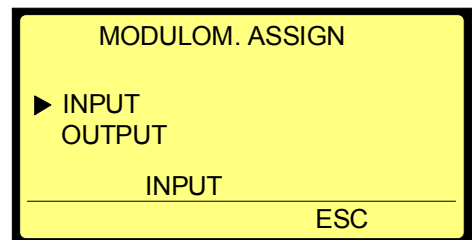


◆ **Modulometer AUTO selection.** The modulometer indication can be selected to be active for the input level or for the output level. In the case of recording a signal, it is advised to set this selection to the input side.

Follow the same rules as illustrated before to find:

“MODULOM. ASSIGN”
“INPUT”

and push



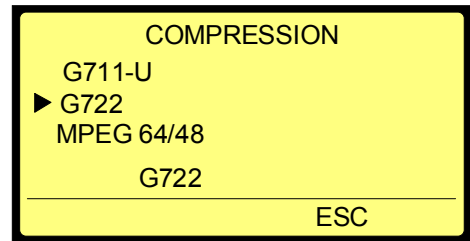
- ◆ **Compression selection G.772.** 12 different kinds of compression can be selected depending on the requirements of the studio or the optional ISDN connection. In this example to simplify matters, we will select the G.722 compression mode which records at a bit rate of 64kb/s with a bandwidth of 7kHz and which can only be used in a mono mode.

Follow the same rules as illustrated before to find:

“COMPRESSION”

“G722”

and push



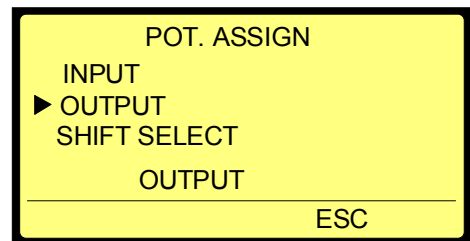
- ◆ **POT. ASSIGN selection.** The next selection that needs to be made is the choice for the “AUX. IN & LINE OUT” potentiometer.

Follow the same rules as illustrated before to find:

“POT. ASSIGN”

“OUTPUT”

and push



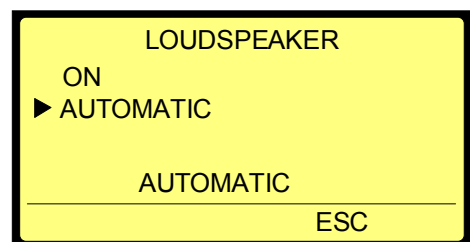
- ◆ **Speaker AUTO selection.** This selection is similar to the modulometer. During record or test, the speaker is automatically switched off and during play or edit, it is switched on.

Follow the same rules as illustrated before to find:

“LOUDSPEAKER”

“AUTOMATIC”

and push

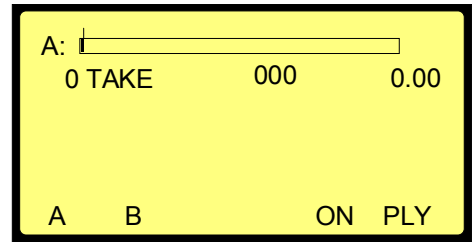


IMPORTANT:

Push several times

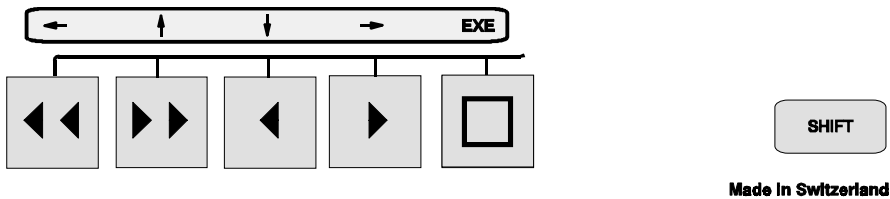


UNTIL



Otherwise "REC" can not be started.

- ◆ **Take number and take length indication.** By keeping the front panel shift key pushed and pushing once the right arrow key, it becomes possible to select 3 possibilities of display indication by using the down arrow key button plus the EXE button. One of them indicates the take number and the corresponding take length during record.



7.3 Settings check.

To check if all the settings just executed are correct, it is easy to recall them by just pushing once the modulometer switch "BATT" a single time and the front display will scroll through them.

The other possibility is to switch off the ARES-C (using the MAIN selector) and switching it back on. This also shows that all settings just made stay in memory even if the machine was switched off.

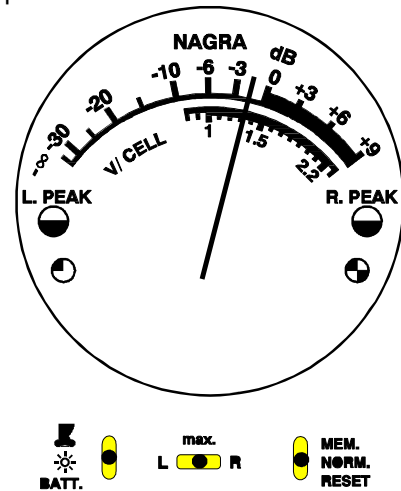
The list appearing on the display will be the following: **G.722**

**ALC. OFF
AUX. OFF
IN LEFT
POT. OUT
LEV. AUTO
SPK. AUTO**

7.4 Microphone level adjustment in test mode.

- ◆ Verify that the "EE, AUTO, TAPE" selector is in the "AUTO" position.
- ◆ Verify first of all that the modulometer selector (below the meter) is in the middle position (max.) or in the left position (L) and the next selector to the right is in the "NORM" position.

- ◆ While speaking into the microphone, adjust the left “MIKE LEVEL” potentiometer until the meter is oscillating close to the 0dB mark. If this is not possible, change the position of the corresponding “SENSITIVITY” (1mV/hPa, 4mV/hPa, 0.2mV/hPa) selector below the potentiometer. **The sector from 0 dB to +9 dB is the headroom area for the AD converter.** This means that the input signal may be increased with max. 9 dB before the AD converter starts to saturate.



7.5 Headphone control.

- ◆ Connect a stereo headphone to the jack connector. Adjust the front AUX. IN & LINE OUT potentiometer to approx. 0dB on the potentiometer scale. While speaking in the microphone, adjust the phones level potentiometer (next to the headphone jack connector) to obtain a correct headphone level. As a G.722 (MONO) compression has been selected earlier, the signal will be audible on both headphone channels.


7.6 Record.

- ◆ At the moment that the modulometer is moving within the corresponding range, all presets are made and the recording can be started. Put the main selector in the “RECORD” position. Automatically the red led on the front display as well as the red led on the top deck will light. This indicates that the ARES-C is in the record mode. The front display shows the number “001” on the left side which indicates the take number and on the right side a counter increments indicating in minutes and seconds the instantaneous length of the recording in progress.
- ◆ To stop the recording, set the main selector back to the “TEST” position. The front display now shows the take number “001” on the left side and the total length of the take 1 on the right side.

7.7 Record with markers.

- ◆ Now that the first recording has been made, let us try a second one in which a marker will be inserted. Set the main selector back to “RECORD” and observe that the take number on the front display now indicates “2” instead of “1”. Simultaneously, on the right side of the front display, the counter restarted from “000”. At the moment that a marker needs to be inserted during record, just push the “ ” button once (below the EXE sign) and automatically take “3” starts. During playback of take “2” and “3” afterwards, no interruption will be detected.

8.0 PLAYING BACK FROM THE FRONT PANEL.

- ◆ During an interview, it happens sometimes that a short playback of the last part of a take or a previous take is asked. This can be done without being obliged to manipulate the editor.
 - ◆ Once the recording has been finished, the main selector can be immediately set to the "PLAY" mode. Instantaneously, the last recorded take will be played back from the beginning (counter 000). By using the function keys, it becomes possible to start over the same take, to skip to previous takes as well as playing back at four times the nominal speed forwards or backwards. This time, the function keys themselves, are active rather than the features indicated in the rectangular box above the keys.
- 
- ◆ The single left arrow button permits, if briefly pushed, to start the play of the same take from the beginning.
 - ◆ The single left arrow button also permits, if pushed twice, to skip to the beginning of the previous take and playing back immediately.
 - ◆ The single right arrow button permits, if pushed briefly, to skip to the next take and play back immediately.
 - ◆ The double right arrow button permits to playback the take at 4 times the nominal speed.
 - ◆ The double left arrow button permits to reverse playback the take at 4 times the nominal speed.
 - ◆ If a "PAUSE" is needed during playback, just put the main selector to the "STB.BY" position or push the "STOP" button once. The playback stops automatically and will restart from the same point once the main selector is set back to the "PLAY" mode or the "STOP" button is pressed again.

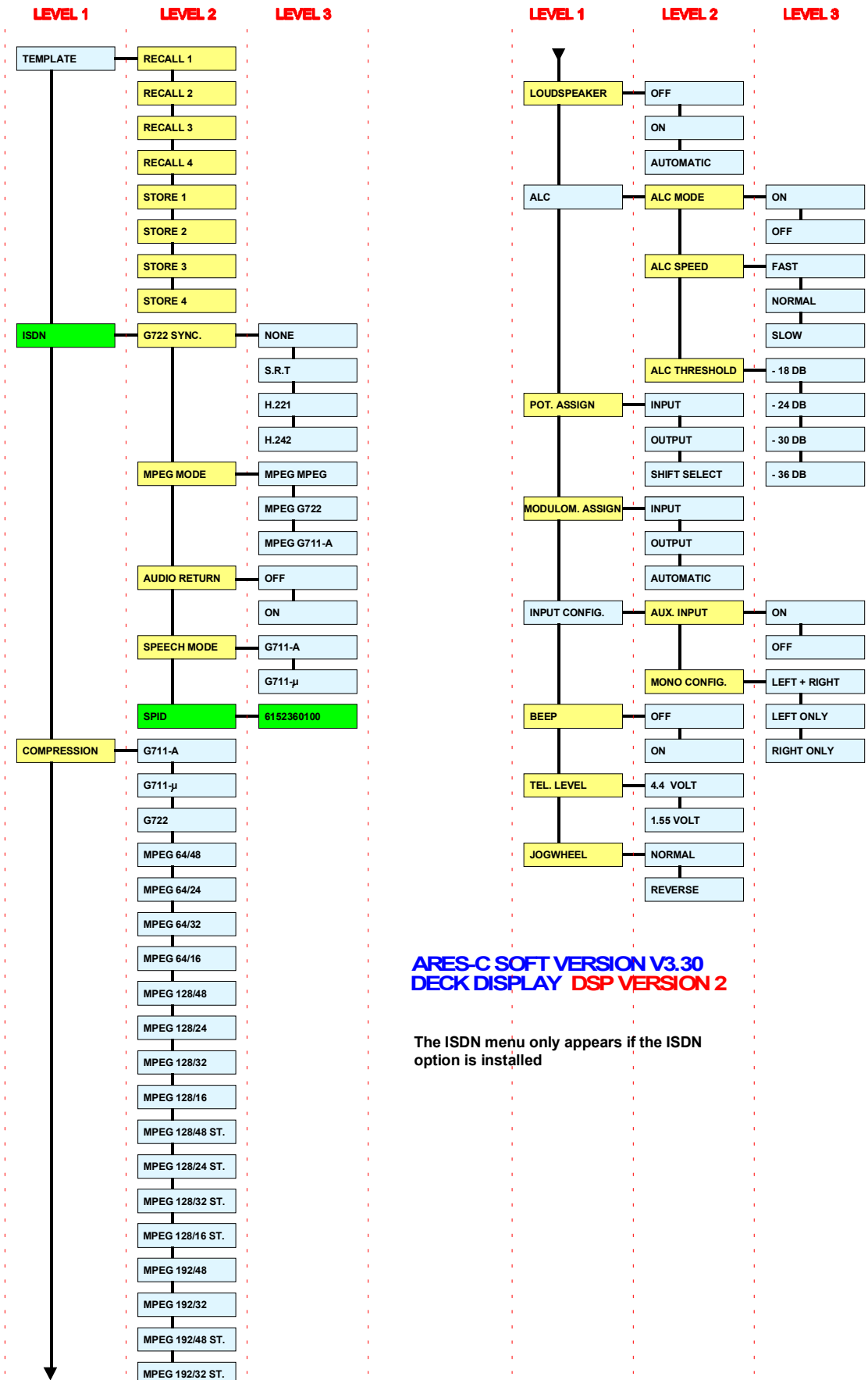
9.0 STARTING THE NEXT RECORDING AFTER PLAYBACK

- ◆ If suddenly during playback of any recorded take, a new recording needs to be started immediately, just put the main selector back to "RECORD" without worrying where the playback was located. No previous recordings can be accidentally erased by doing this. The new recording will start from the end of the last recorded take located on the card.

CHAPTER II - IN THE FIELD OPERATION

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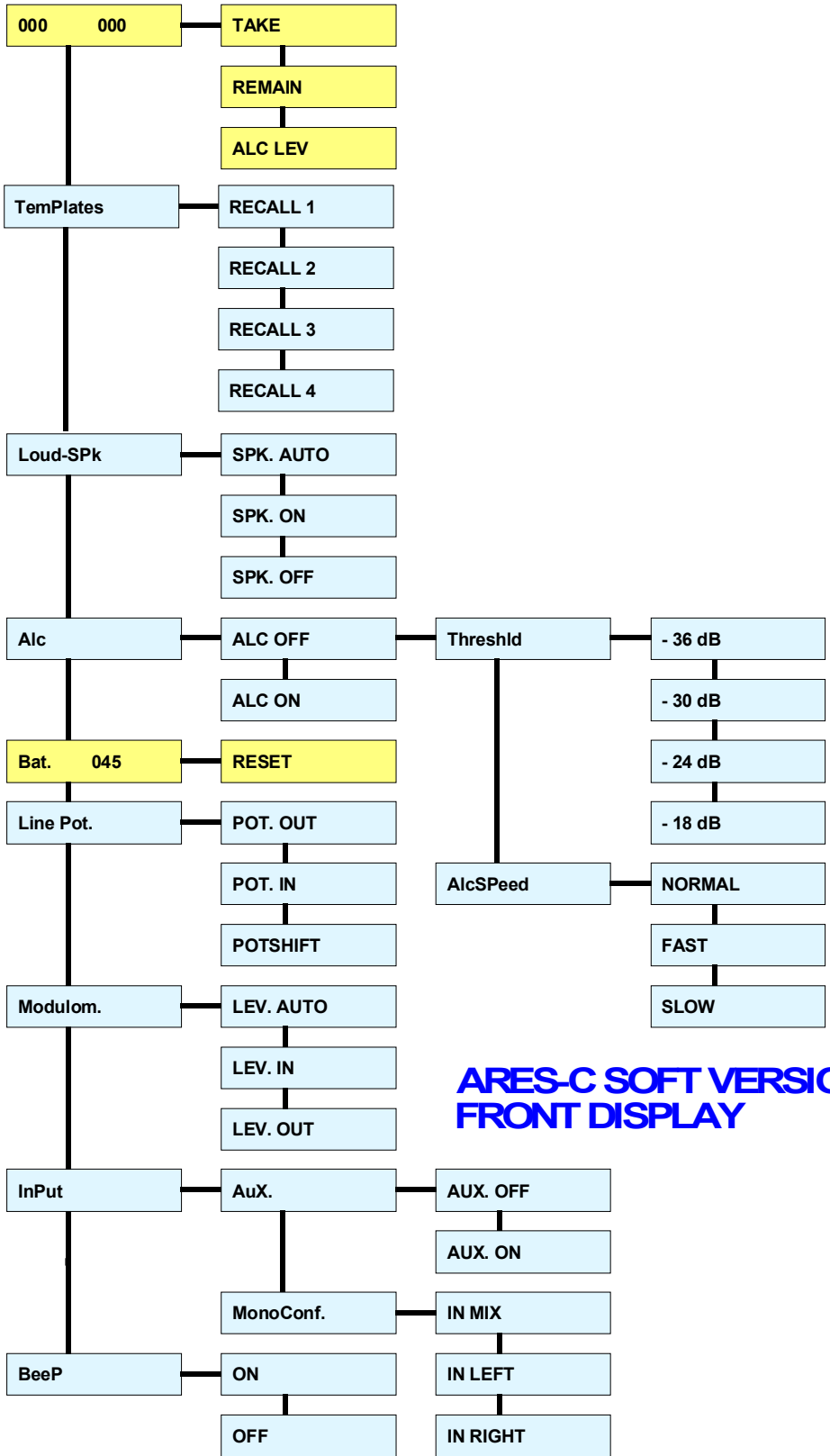
1.0 TREE DECK DISPLAY



**ARES-C SOFT VERSION V3.30
DECK DISPLAY DSP VERSION 2**

The ISDN menu only appears if the ISDN option is installed

TREE FRONT DISPLAY





**ARES-C SOFT VERSION V3.30
FRONT DISPLAY**

2.0 MENU MODE FRONT DISPLAY

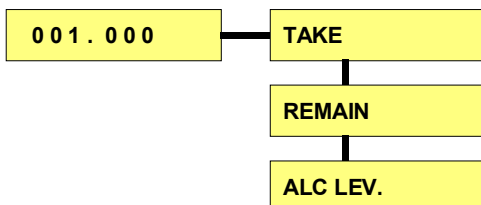
The ARES-C incorporates a system of menus similar to the "tree" of directories found on the hard disk of a PC. The functions, accessible through the menu mode, are operations that are not performed too frequently during normal operation of the machine.


The menus are displayed on the LCD display on the front panel or the deck of the machine. They can be accessed at any time except when the main function selector is in the STOP position. The five push-button transport keys on the front panel are used to move around the menus while the shift button is pressed.

The menu mode is turned on by holding down the SHIFT key AND one of the three arrow keys ("↑", "↓" or "→") which are the functions written above the  respectively.  The key (or STOP in earlier versions) is used for EXECUTE to activate a feature on the display, which will be confirmed to the user by a beep in the headphones and on the internal loudspeaker (if BEEP is selected to ON). The arrow keys can be used to move UP, DOWN, LEFT and RIGHT through the tree. A full description of each of the menus follows.

Shift key must be held down while going through menus:

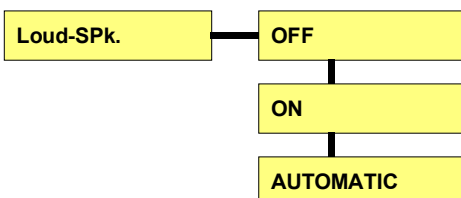
2.1 DISPLAY MENU



The display menu allows the operator to select the default display to be indicated on the front panel of the machine. The factory default display is the "TAKE" as shown above which indicates the take # followed by the time from the start of the take in minutes and seconds (similar to the track # and time display of a domestic CD player). Move to the right then down using the arrow keys on the front panel and then select by pressing EXE (Shift + ) , (or Shift + STOP on older versions)

- TAKE Will display the three digit take # on the left side of the display and the time from the start of the take on the right side, indicated in minutes and seconds.
- REMAIN Will display the remaining recording time available on the PCMCIA card presently in use. It is indicated in minutes.
- ALC LEV. Will display the compression level of the ALC. The display shows an "<" on the left side followed by zero's with crosses in them depending on the amount of compression. When the ALC is at maximum a reverse ">" is indicated on the right side of the display. The number of zero's displayed indicates in dB the amount of compression. Please refer to appendice 1 for a more detailed explanation of the theory of operation of the ALC.

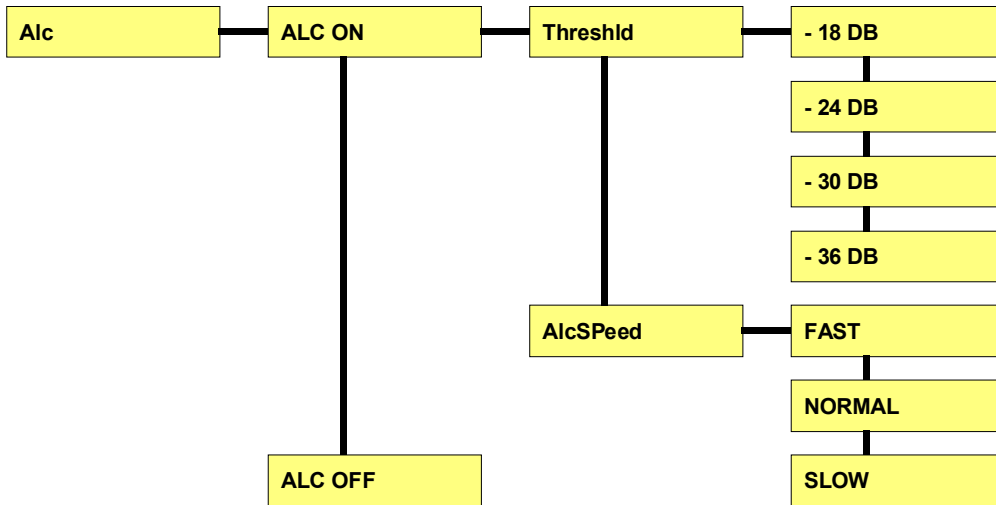
2.2 LOUDSPEAKER MENU



This menu allows the user to select the operating mode of the internal loudspeaker. It can be turned ON and OFF by executing the desired mode or it can be in the AUTO mode. In AUTO the internal loudspeaker will be

OFF during TEST, RECORD and TRM (transmit) modes and will be ON during PLAY and STD.BY (old keyboard version: EDIT) modes.

2.3 A.L.C. MENU



This menu selection allows the Automatic Level Control circuitry to be turned ON or OFF as well as the operating mode / parameters of the internal ALC circuitry. Recordings may also be made without the use of ALC by simply putting the main function selector to the second record position. If the ALC is selected to OFF then both record positions of the main selector are the same. If it is ON then the ALC circuit will also be activated when the main function selector is in the TEST position. When the ALC is activated the amount of compression can be seen on the display on the front panel and is indicated by zero's with crosses in them.

The THRESHOLD position gives the user access to four possible settings, from where the ALC will activate. The ALC speed position lets the user select one of three different speeds of ALC reaction. The general operating position is NORMAL which corresponds to a hold of 2 seconds and then a fall off of the ALC during approx 6 seconds. (This is the same as the time for the NAGRA IS). The slow position corresponds to a hold of 2 seconds followed by a fall off during approx. 30 seconds and the FAST position corresponds to a hold of about 0.4 seconds and a fall off during 1.2 seconds.

For a more detailed explanation of ALC operation please refer to appendice 1 of this manual. The ALC is not active if the AUX input is selected.

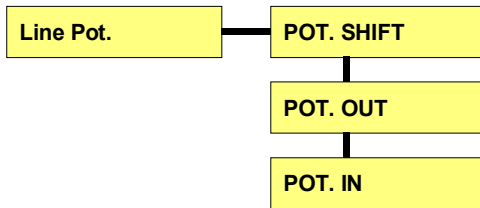
2.4 BATTERY MENU



This position allows the user to see how long in hours and minutes the ARES-C has been powered up since the last time counter was reset. This will give the user an idea of how much more recording time is left in a set of batteries. Pressing the right arrow will display RESET and then pressing EXE will reset the counter to zero.

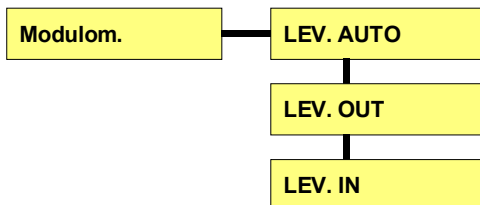
This is not reset automatically when the battery box is changed and must be done by the user.

2.5 LINE POT MENU



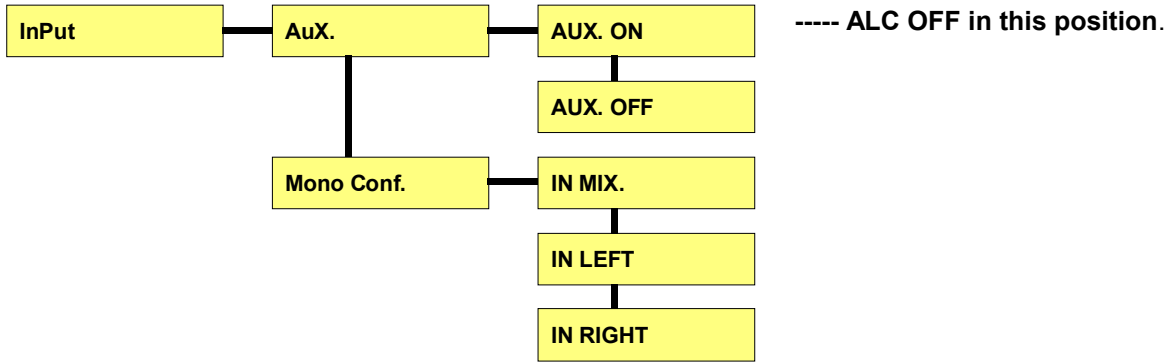
This menu gives access to the mode of operation of the third potentiometer on the front panel of the machine (AUX IN & LINE OUT). The line potentiometer of the ARES-C can be set to the POT.SHIFT position. That is to say that it will adjust the input signal if the SHIFT key is kept pressed and it will adjust the OUTPUT signal if the SHIFT key is not pressed. It is also possible to select either the AUX. input (POT. IN) or LINE output (POT: OUT) in the menu mode and it will only adjust this position irrespective of the selected operating mode of the machine. When changing this selection, the previous pot position will be kept in memory even if the machine is switched off, assuming there is power available.

2.6 MODULOMETER MENU



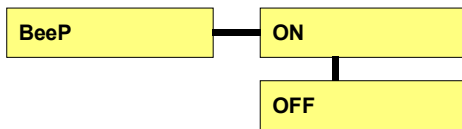
The modulometer of the ARES-C can be set to the AUTO position in which case it will behave as on a standard NAGRA. That is to say in TEST and REC it will indicate the input signal and in playback it will indicate the OUTPUT signal. It is also possible to select either LEVEL IN or LEVEL OUT in the menu mode and it will only monitor this position irrespective of the selected operating mode of the machine.

2.7 INPUT MENU



The input menu allows the configuration of the various inputs of the ARES-C. The MONO CONFIGURATION position allows the two microphone inputs to be mixed together or only one or other to be connected. This selection was necessary to cancel the noise generated by an unloaded mic pre-amplifier especially when the ALC is not activated when only one microphone is being used. The AUXILIARY position permits the third input, located on the 15 pole "D" connector, to be turned on. If it is turned on then it will be mixed with the signal coming from the left and right mic pre-amplifiers.

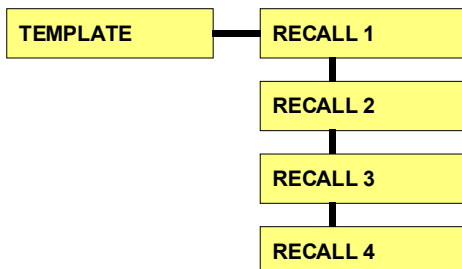
2.8 BEEP MENU



The features covered are the BEEP function which can be turned ON or OFF. When the beep is ON it will sound in both the headphones and the loudspeaker providing it has been selected. The possible beep signals are:

- | | |
|---------|---|
| 1 Beep | Execute of a function in the menu mode (function accepted)
Error message has been displayed on the front panel display
Only 1 minute of recording time remains on the card presently being used |
| 2 Beeps | Low Batteries has been detected
Function execute refused in the menu mode |
| 3 Beeps | DSP is not functional anymore |
| 4 Beeps | Internal settings have been lost (SET LOST on front display) |

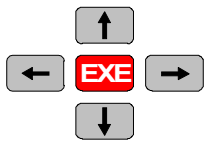
2.9 TEMPLATES



Any blue menu from level 1, level 2 or level 3 (see tree page 2) can be stored and recalled from the memory of the machine. Maximum four total different configurations can be selected from the front display.

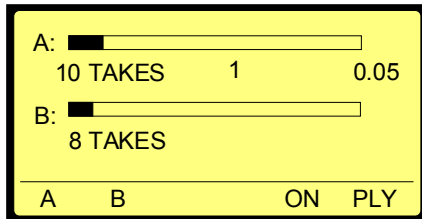
3.0 MENU MODE ON DECK

The functions, accessible through the menu mode, are operations that are not performed too frequently during normal operation of the machine.

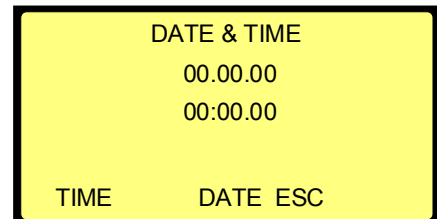


Accepted executions will be confirmed to the user by a beep in the headphones. To access to the menu, once the "ON" button (F4) is pressed, the "SET" function button (F3) must be pressed. The arrow keys can be used to move UP, DOWN, LEFT and RIGHT through the tree. The execute button is the middle one. A full description of each of the menus follows:

3.1 DISPLAY CONTRAST



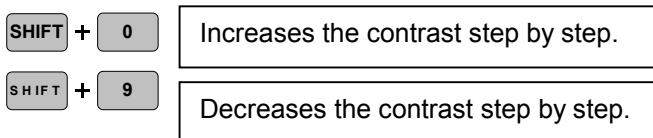
OR



One of these menus appears when the NAGRA ARES-C (having two PCMCIA cards in the slot) is switched on.

Due to memory lost, the right menu appears instead of the left menu. This happens only if the NAGRA ARES-C was disconnected from the battery box for several hours. Push the "ESC" button to return to the left display.

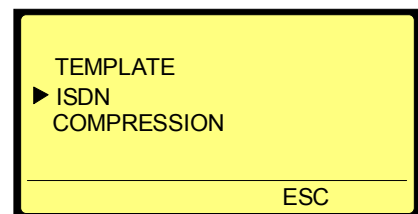
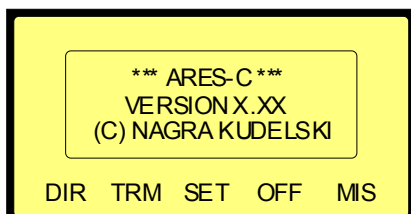
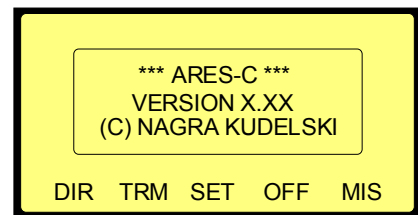
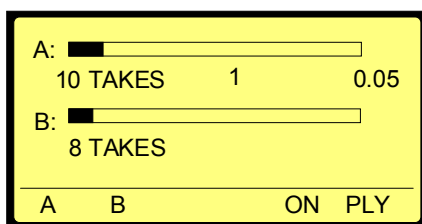
If the contrast of the screen is not correct, the following adjustment needs to be done:



The final result once the "SHIFT" button released is stored in the E² prom of the ARES-C.

To enter into the settings of the menu, the "ON" button (F4) followed by the "SET" button (F3) must be pressed.

3.2 SETTINGS MENU

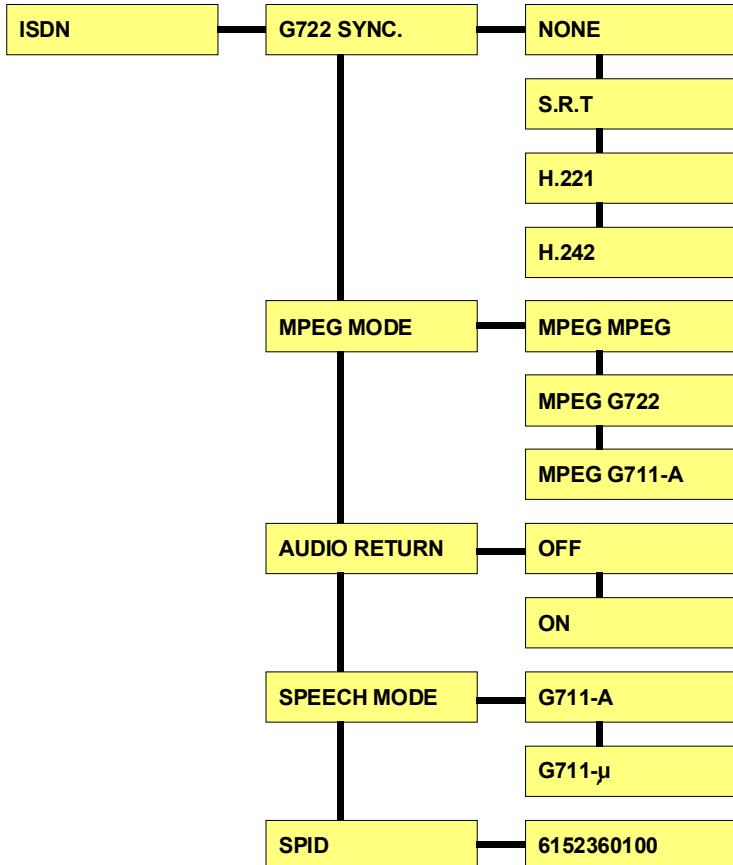


3.3 ISDN MENU

This menu only appears if the ISDN option is installed.



By using the "UP" & "DOWN" arrow buttons, scroll to find the ISDN menu. Once "ISDN" is in front of the marker, push the "RIGHT" arrow button once. The next screen on explains first of all that the "ISDN" menu has been selected (ISDN on top of the screen). By using the "UP" & "DOWN" arrow buttons, a second level selection can be made between "G722 SYNC." and "SPID". The complete ISDN tree is as follows:



The G.722 sync. selection can be "NONE", "S.R.T.", "H.221", or "H.242". This depends on which kind of codec is used on the other side of the ISDN line and is explained in detail in chapter IV.

The "MPEG MODE" selection gives the possibility to set-up the full duplex algorithm for MPEG MPEG both ways, or MPEG G722 or MPEG G711-A. See explanation in chapter IV.

Permits on the headphone and speaker to turn ON or OFF, the return signal from the line and microphone input as well as the playback from the card during an ISDN connection.

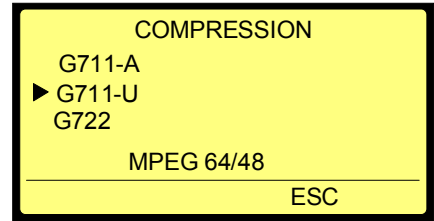
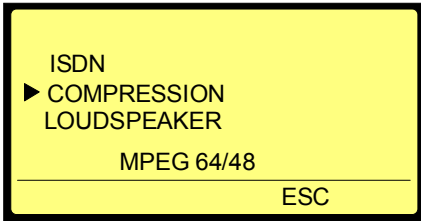
The "ISDN SPEECH MODE" in the transmission menu can be set to G.711-A or G.711-μ law.

The "SPID" menu is only used in the USA. For any other country, the entry must stay blank.


Extended explanations are made in chapter IV. To "EXECUTE" a selection push

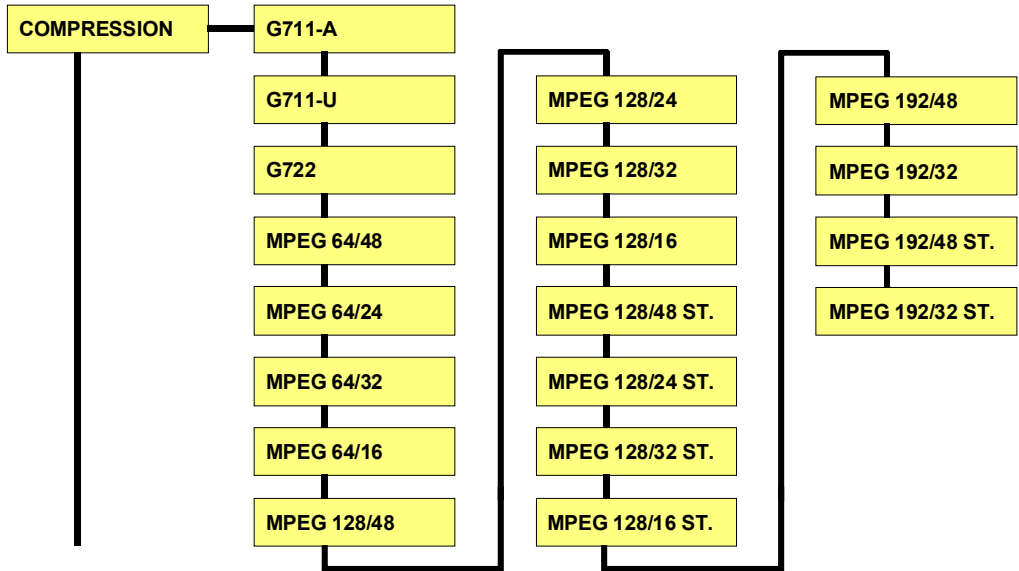


3.4 COMPRESSION MENU

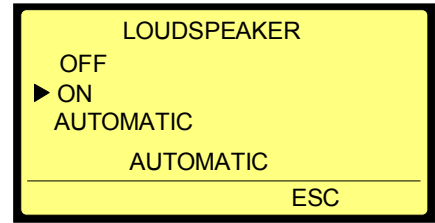
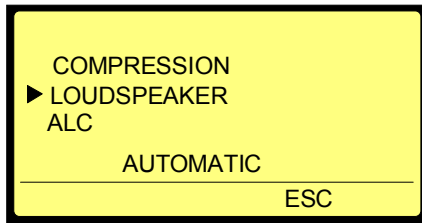


19 different compression algorithms can be selected. The first two are used for "speech" transmission (G.711). All the others are only working in "data" transmission. To "EXECUTE"

Push: 



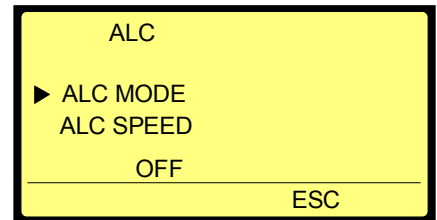
3.5 LOUDSPEAKER



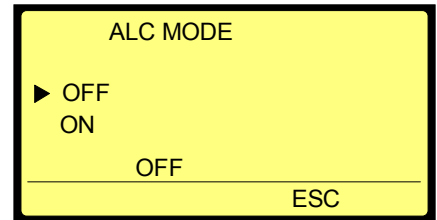
To "EXECUTE" a selection push

This menu selection allows the speaker to be: Always switched ON
 Always switched OFF
 Or automatic switching; ON during EDIT or PLAY
 OFF during RECORD or TEST

3.6 ALC MENU

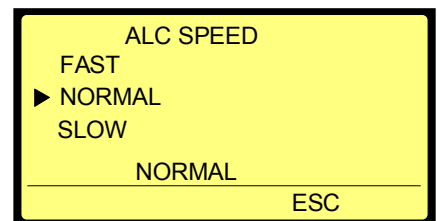
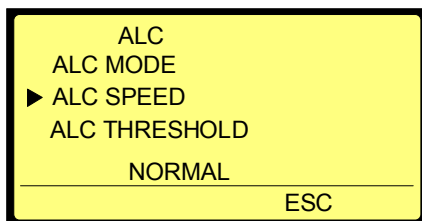


3.6.1 ALC MODE



This menu selection allows the Automatic Level Control circuitry to be turned ON or OFF as well as the operating mode / parameters of the internal ALC circuitry.

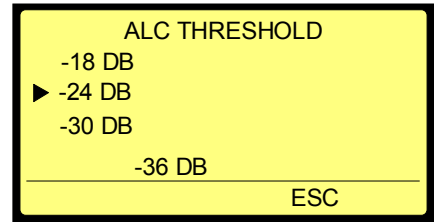
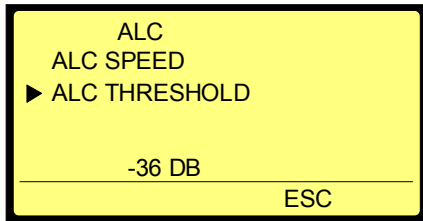
3.6.2 ALC SPEED



The ALC speed position lets the user select one of three different speeds of ALC reaction. The general operating position is NORMAL which corresponds to a hold of 2 seconds and then a fall off of the ALC during approx 6 seconds. (This is the same as the NAGRA IS). The slow position corresponds to a hold of 2 seconds followed by a fall off during approx. 30 seconds and the FAST position corresponds to a hold of about 0.4 seconds and a fall off during 1.2 seconds. For a more detailed explanation of ALC operation please refer to appendice 1 of this manual.

The ALC is not active if the AUX input is selected.

3.6.3 ALC THRESHOLD

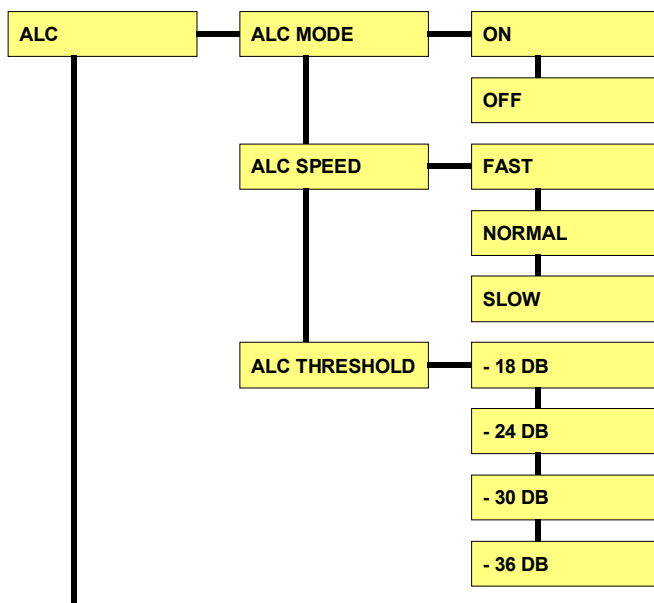


The ALC TRESHOLD gives 4 threshold level possibilities.

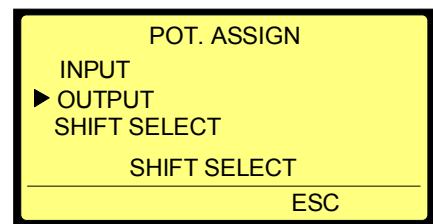
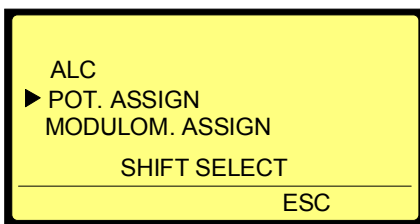
To "EXECUTE" a selection push



Full ALC menu

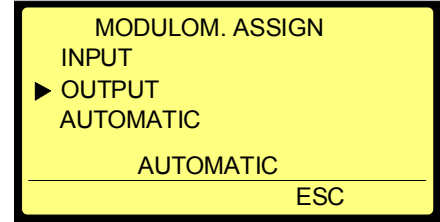
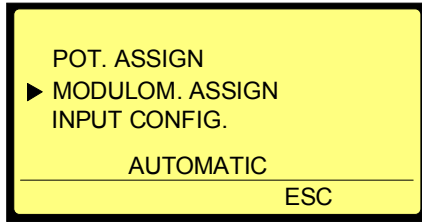



3.7 AUXILIARY INPUT AND LINE OUTPUT POTENTIOMETER



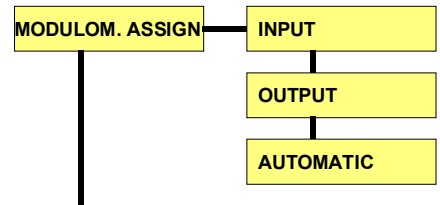
This menu selection gives the possibility to select the use of the "AUX. IN & LINE OUT" potentiometer. If "SHIFT SELECT" has been chosen, the potentiometer will adjust the input level if the front "SHIFT" button is not pressed and it will adjust the line output level if the "SHIFT" button is pressed. The previous adjustment is memorised.

3.8 MODULOMETER ASSIGN MENU

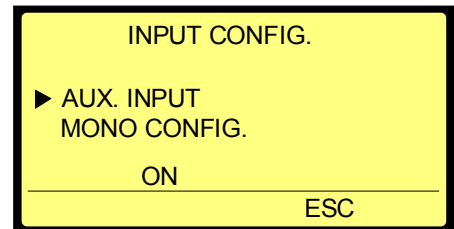
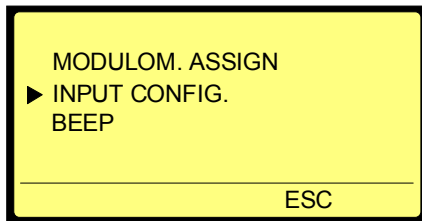


To "EXECUTE" a selection push 

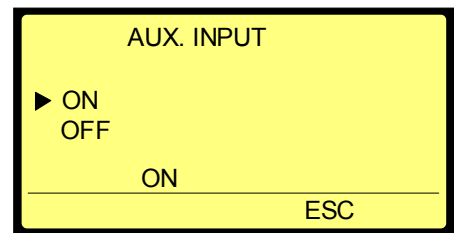
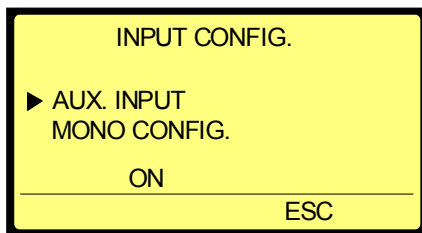
The modulometer of the NAGRA ARES-C can be set to the AUTO position in which case it will behave as on a standard NAGRA. That is to say in RECORD it will indicate the input signal and in PLAYBACK it will indicate the OUTPUT signal.




3.9 INPUT CONFIGURATION MENU



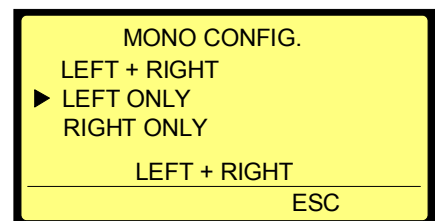
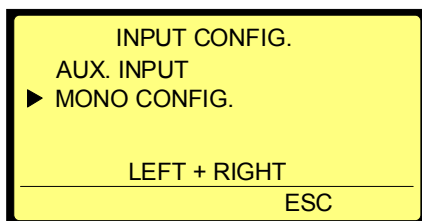
3.9.1 AUXILIARY INPUT MENU



The AUXILIARY position permits the third and fourth input, located on the left side of the NAGRA ARES-C, to be turned on. If it is turned on then it will be mixed with the signal coming from the left and right mic. pre-amplifiers.

To "EXECUTE" a selection push 

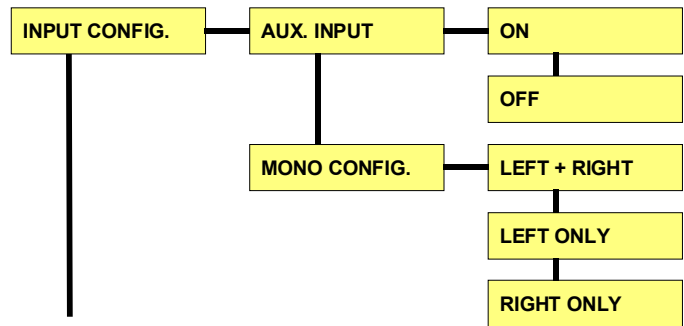
3.9.2 MONO CONFIGURATION MENU



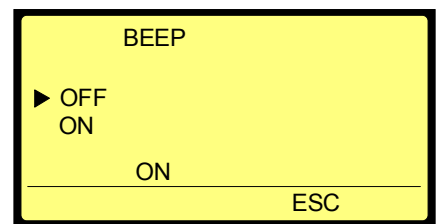
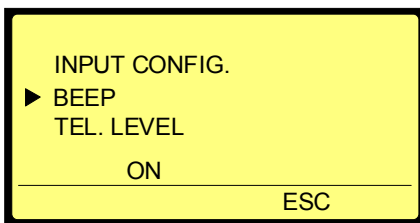
The MONO CONFIGURATION position allows the two microphone inputs to be mixed together or

only one or other to be connected. This selection was necessary to cancel the noise generated by an unloaded mic pre-amplifier especially when the ALC is not activated when only one microphone is being used.

To "EXECUTE" a selection push



3.10 BEEP MENU



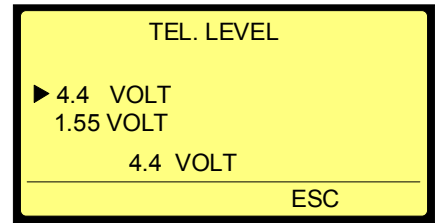
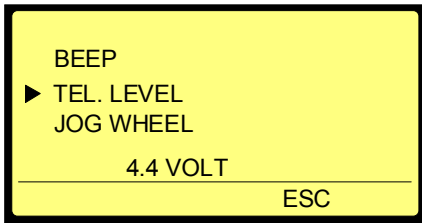
To "EXECUTE" a selection push



When the beep is ON it will sound in the headphones. The possible beep signals are:

- | | |
|---------|---|
| 1 Beep | Execute of a function in the menu mode (function accepted)
Error message has been displayed.
Only 1 minute of recording time remains on the card presently being used |
| 2 Beeps | Function execute refused in the menu mode |
| 3 Beeps | The DSP is not functional anymore |
| 4 Beeps | Internal settings have been lost (SET LOST on front display) |

3.11 TELEPHONE LEVEL

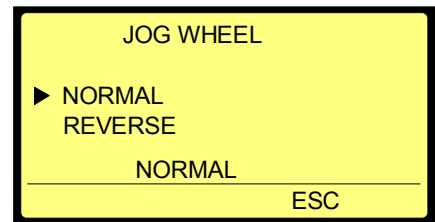
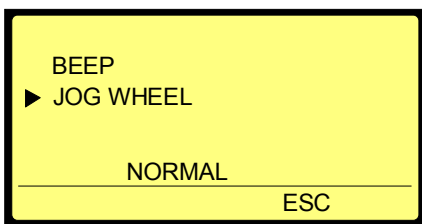


To "EXECUTE" a selection push



This menu selection permits to select the banana output level.

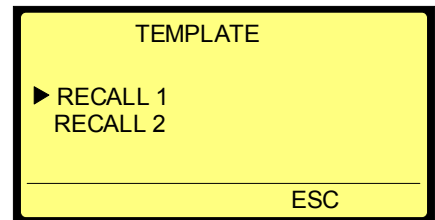
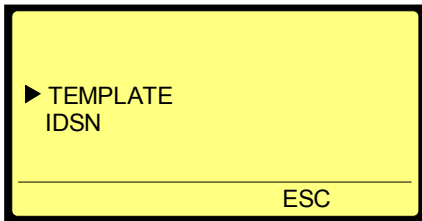
3.12 JOG WHEEL MENU



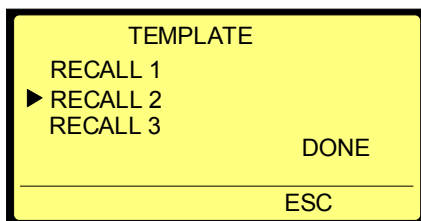
The JOG wheel setting allows the operator to configure the jog wheel according to their personal taste. This setting affects the direction of the tape movement on the display when using the internal editor.

3.13 TEMPLATES

3.13.1 RECALL



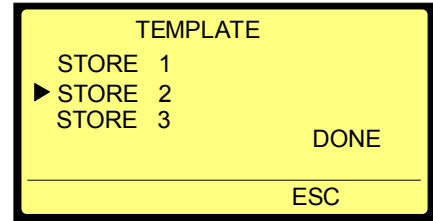
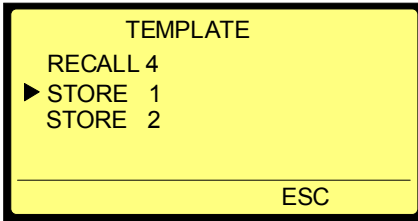
Select the appropriate template by using the up and down arrow keys. Once the EXE key is pressed "DONE" will appear for a few seconds:



ATTENTION:

If nothing was stored in one of the four selected templates, the message "EMPTY" will appear instead of "DONE"

3.13.2 STORE



Select the appropriate template for storage. All the settings executed in the blue menus (level 1, 2 or 3) will be automatically stored in the memory of the machine.

Example after having stored two different configurations:

RECALL 1 = G722 SYNC. S.R.T, COMPRESSION G.722 SPK. AUTO, ALC OFF, POT. OUT, LEV. AUTO, AUX. ON, LEFT ONLY, BEEP ON, 4.4 VOLT, NORMAL.

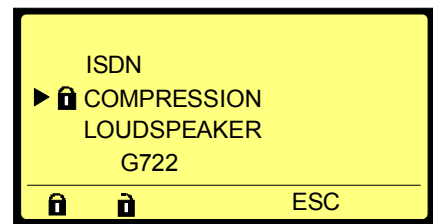
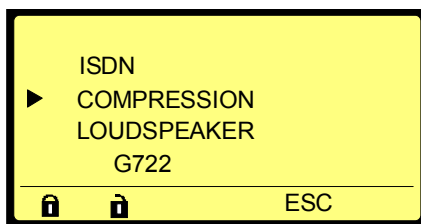
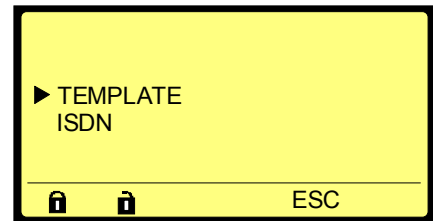
RECALL 2 = MPEG 128/64 ST, SPK. OFF, ALC ON, ALC FAST, -18DB, POT. SHIFT SELECT, AUX. OFF, BEEP OFF, 1.55 VOLT, REVERSE.

3.14 PASSWORD

By introducing a password, an additional menu will appear. The password (*****) needs to be introduced once the "SET" function key is pressed:

The F1 function key shows a closed padlock and the F2 function key shows an open padlock. With this feature, any of the yellow menus from level 1, 2 or 3 can be locked (see tree page 2). The only exemption concerns the SPID in the ISDN domain where the journalist must be able to modify this number when he is travelling in the US.

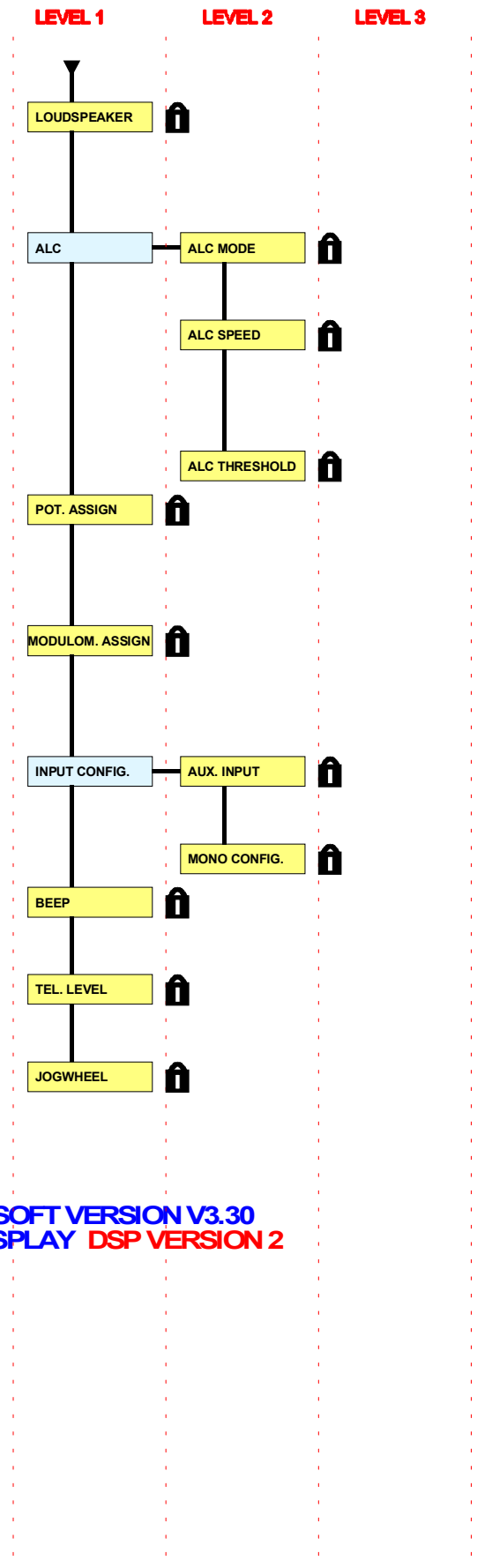
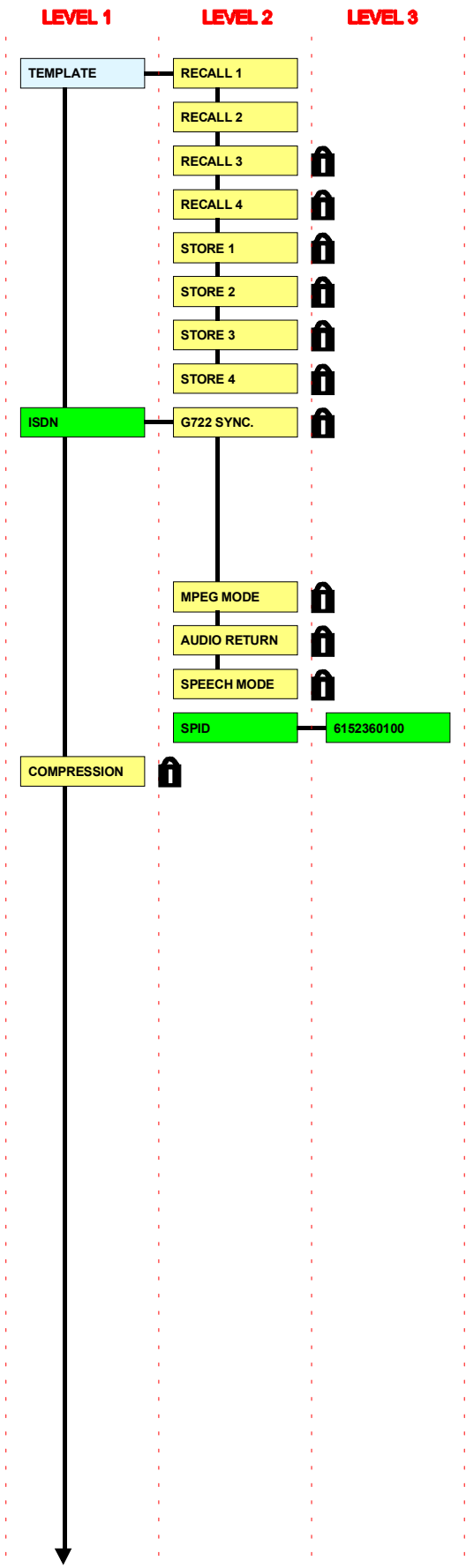
Example: Locking the compression menu:



The closed padlock appears in front of the compression menu. As long as the password is not disabled, it's still possible to navigate in the compression menu to change the settings.

To disable the password, the machine must be switched off once.

The next example shows an ARES-C tree with a maximum of menus locked. Only the "RECALL1" is accessible. This is important in the case if the machine has lost his memory (no batteries for a few hours). Automatically after a power ON, the machine will select the settings of the first not locked "RECALL" menu. If all "RECALL" menus are locked, the machine after a memory lost will take the initial standard factory settings.



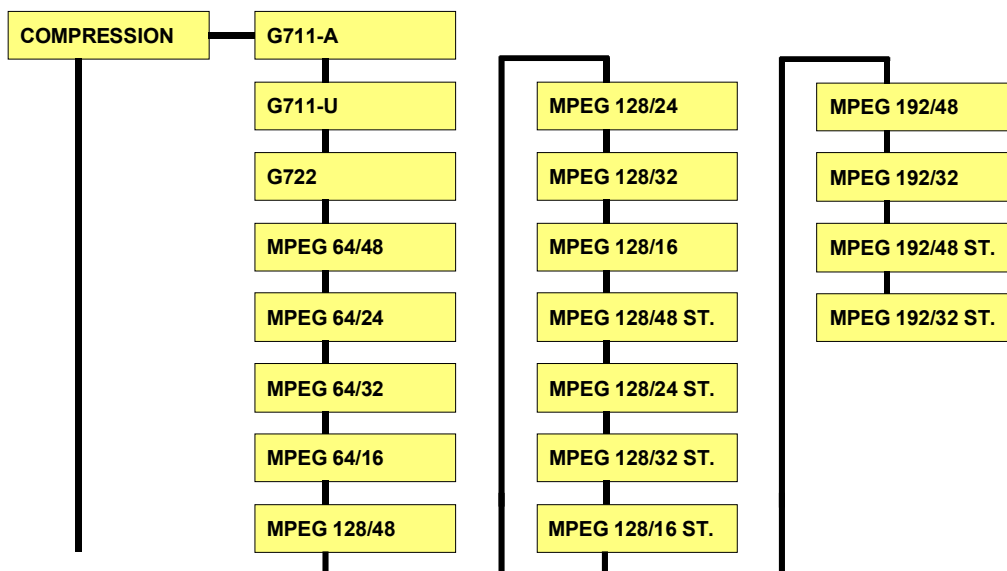
4.0 HEADPHONE MONITORING

Headphone monitoring is possible at all times, during record, playback and editing. Simply connect the headphones to the Jack connector and adjust the desired monitoring level using the potentiometer # 5. It should be noted that if the line output potentiometer # 10 is at minimum then there would be no headphone output as the two potentiometers are interdependent.

5.0 COMPRESSION SELECTION

The ARES-C can operate using four different main audio compression systems, each of which will give different operating performances of the machine. The four possible systems are:

- G 711-A This compression system is similar to the μ -LAW system but is used in the rest of the world other than the USA, having a 3.5 kHz bandwidth and 8 kHz sampling frequency.
- G 711-U This selection is the compression system generally used in the U.S.A. and will allow a bandwidth of 3.5 kHz. In this position the digital sampling frequency is 8 kHz.
- G 722 G 722 was developed after μ -LAW and A-LAW and is now becoming the world standard compression system for voice compression. It offers a bandwidth of 7.5 kHz and in this case the sampling frequency is of 16 kHz.
- MPEG This position corresponds to the MPEG-1 Layer II compression norm. Various different modes of using MPEG compression on the ARES-C are available depending on the application. Different bit rates and different sampling frequencies can be selected and naturally bandwidth will be different according to the mode selected. The bit rate can be selected to be either 64, 128 or 192 kbits per second. This is an important selection as it depends whether the ISDN output is to be used. The ARES-C can only operate on 1 "B" channel which requires the use of 64 kbits/sec. In order to use 128 kb/s two "B" channels on the ISDN network would be needed. (Please refer to the appendice to this manual for more information regarding ISDN). Once the bit rate selection has been made the sampling frequency can be chosen.



The choice of the compression method to be used for a particular recording can be made by the operator by means of the MENU mode. The user should select the corresponding compression mode according to the job in hand. It is important that the correct compression method is selected BEFORE the recording is made as it is not possible to record using one method and then play back using another. Once a compression mode has been selected the machine will stay in this mode until a different mode is selected.

The machine will automatically play back all the takes in the directory listing that were recorded using the compression mode presently selected. If for example the machine is set to G.722 compression and the user tries to play back a take recorded in MPEG then the machine selects the corresponding compression and plays back only that selected file.

If the machine is put back into the record mode, it will go back to the initial settings. In this case G.722

If the main selector is set to the PLAY position then only the takes recorded with the current compression selection will be played back. All other recordings on the card will be "skipped".

In order to see the compression mode used for any specific recording, turn on the internal editor and select DIR to put the directory on to the display, and then by means of the RIGHT ARROW key scroll the display to the right until the compression mode is displayed. Such a display will look something like the following:

```
A: 001      0:04
000  FORMAT
▶ 001  G.722
002  G.722
003  MPEG 64/48
END
EDT DEL A-B ESC PLY
```

Important: It is not possible to change the selected compression of the ARES-C while either the machine is in RECORD or while the internal editor is turned ON.

6.0 DIRECTORY RECORDING

The ARES-C records a "DIRECTORY" automatically when the card is initially formatted. As recordings are made or edited the directory is automatically updated and is stored in the card. A directory of a card can contain up to 408 "takes" maximum. If the user tries to switch the machine to record and there are already 408 takes on the card then the red LED will not light and the recording will not take place. (if the second slot has a card inserted and it has less than 408 takes already recorded then the recording will be made on the other card).

If the user tries to EDIT a take on a card that has 408 takes already recorded then the message DIRECTORY FULL will appear on the display when the EDT key is pressed.

7.0 MAKING THE RECORDING

USING MICROPHONES

The procedure described below covers the important steps necessary in order to record using the ARES-C. It does not give indications as to the type of microphones or their placement but purely the steps that should be followed.

Initially ensure that the machine is either connected to an external power source or that internal batteries are fitted into the battery compartment. Install a PCMCIA FLASH memory card into one of the two slots provided on the top deck of the machine. If the card is new then format the card as described in the first chapter of this manual under the section **QUICK START, "6.0 Installing the flash card and formatting"** or **CHAPTER I "3.1 Formatting a PCMCIA card"**.

Select the compression mode to be used in the menu mode. (default setting is G722), as explained earlier in this chapter. If a joint stereo recording in MPEG or a mono MIX recording in G722 is to be made then two microphones will be needed.

Connect the microphone(s) to the input connectors on the left-hand side of the machine. Turn the rotary powering selectors above the connectors to the position according to the microphones being used. Generally reporters microphones will be of the DYNAMIC type and require no powering. If in doubt, contact the microphone manufacturer for details. Once this is done set the microphone sensitivity selectors, located under the microphone potentiometers on the front panel according to the type of microphone being used. The sensitivity selections for a specific microphone are indicated in the documentation of the microphone.

At this point the selection of one of the three filters can be made if required. The three filters are LFA (Low Frequency Attenuation), SPEECH and FLAT, the curves for these are all indicated in chapter I of this manual.

Set the main function selector of the ARES-C to the TEST position. Set the EE / AUTO / TAPE selector to the AUTO position.

Turn the main function selector switch to the TEST position. The display on the front panel of the machine will immediately scroll through the current settings of the machine and will end up in the "TAKE / TIME" display. Connect a pair of headphones to the headphone output connector on the right-hand side of the machine and then speak into the microphone. Adjust the microphone input potentiometer (Left and / or Right) on the front panel until an indication is shown on the modulometer. If no indication is shown when speaking into the microphone refer to chapter V "Problem Solving" of this manual. Adjust the headphone level using the headphone output level potentiometer located to the right of the headphone connector. Check also that the AUX. IN / LINE OUT potentiometer is not in the fully counter clockwise position. If no signal is heard in the headphones refer to chapter V of this manual.

Once the sound is heard in the headphones and is correctly indicated on the modulometer simply move the main function selector to one of the two RECORD positions. The red LED on the front panel, as well as the "W" LED corresponding to the card in use, should both light up. The first of the two record positions will activate the ALC circuitry. Please refer to the annex of this manual for a more detailed explanation of the ALC circuitry and its operation. Pressing the STOP "■" key at any time during the recording will cause a new take to be created in the directory at that point. This is very useful for locating special events after the recording has taken place. When the recording is completed return the main function selector to the STOP position.

CHAPTER III EDITING

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1.0 INTRODUCTION

The ARES-C is fitted with an internal editor allowing features such as Cut, Paste, Copy etc. The contents of the PCMCIA card are listed in a DIRECTORY which indicates:

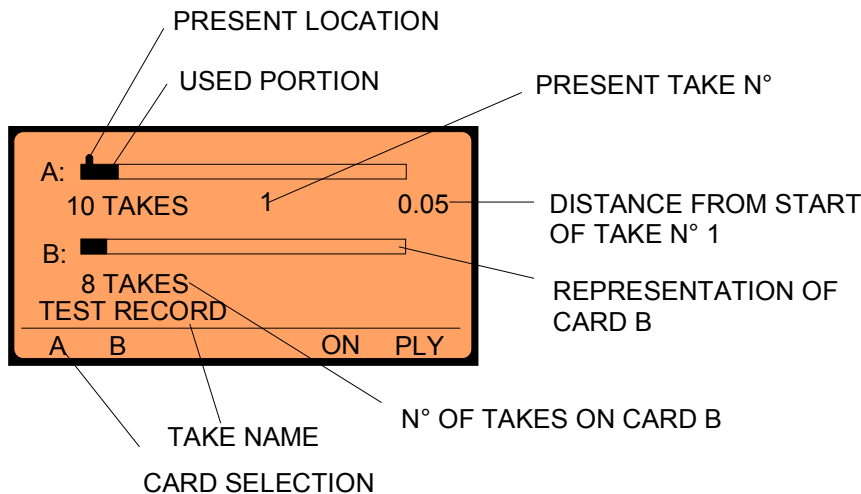
- The card presently being used
- The number of takes
- The duration of each take
- The time and date of the recording (according to the RTC)
- The compression mode used for the recording

It is then possible to switch on the built-in editor and make an EDL using IN and OUT points as desired. The final edited information can then be saved on the card, copied either to another PCMCIA card, or to the various OUTPUTS of the machine (AES, Telephone or ISDN if the option is installed).

2.0 TURNING ON THE EDITOR

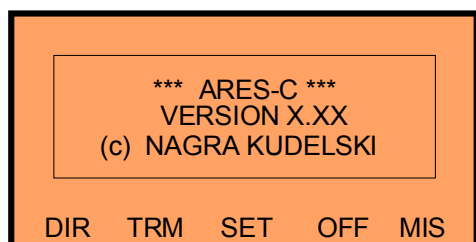
Make sure that a PCMCIA card is installed in either slot A or slot B on the top deck of the machine. Put the main function selector on the front panel of the machine to the EDIT / STD BY position. Be sure that the compression mode of the ARES-C is set to the same mode as the compression mode of the takes in the card that are to be edited. From this point onwards, all edit operations are performed with the buttons on the top deck of the ARES-C. Once the main selector has been set to the EDIT / STD BY position then the graphic LCD display will immediately show the graphic representation of a "tape" corresponding to the data that is recorded in the PCMCIA card(s) - as it does during normal recording or playback. It also shows the number of "takes", the present take being read and the distance from the start of the take in minutes and seconds.

Aligned with the function keys there are several commands written on the bottom portion of the display. At this point the editor is still not ON we can refer to this display as the main display.



Pressing F1 "A" or F2 "B" selects the active PCMCIA card. Only the information corresponding to the inserted cards is displayed. The drawing above is an example of a machine which has two PCMCIA cards installed.

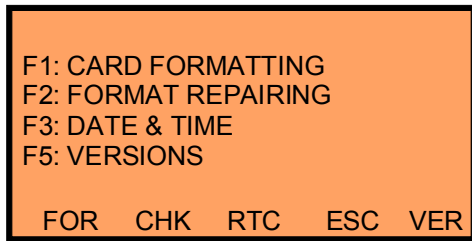
In order to switch on the internal editor of the ARES-C press the F4 "ON" key. The display will now show the software version installed in the machine. F4 immediately toggles to become the OFF key which is used to turn the editor OFF again.



From this screen the user has five choices: "DIR" (directory) by pressing F1, "TRM" (transmit) by pressing F2, "SET" by pressing F3 to change the settings of the machine, "OFF" to turn the editor off using F4 or "MIS" (miscellaneous) by pressing F5. DIR will display the directory on the selected card (covered later), TRM will also be covered in more detail later.

3.0 MISCELLANEOUS SETTINGS "MIS"

MIS gives access to various features. Pressing F5 (MIS) the display will change to the following:



From this screen there are five choices. "FOR" (F1) is used to FORMAT a new PCMCIA card which is explained in detail in chapter 1 of this manual.

4.0 CARD CHECKING "CHK"

It is possible that an error may be caused during the recording of a card, for example if the card is removed accidentally or the power is suddenly disconnected while data is being recorded onto the card. When the machine is turned on again the message "FORMAT CORRUPTED" will appear on the display. To correct such problems, turn on the editor, and press F5 "MIS" followed by F2 "CHK" (CHECK) and the machine will immediately indicate the type of error that has occurred. The possible messages that will be displayed will be:

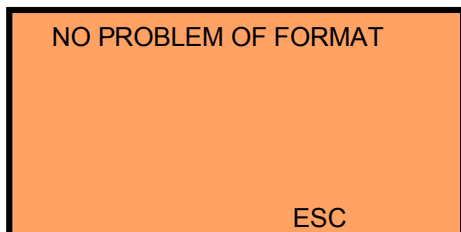
- INVALID ENTRY
REFORMAT CARD

In this case the only remedy for the problem is to completely reformat the card.

- UNCLOSED FILE
RECOVER ?

In this case using the card check (CHK) menu of the display the remainder of the card will be filled. The last take on the card which will fill the card. Once the RECOVER command has been activated, all the takes, including the unclosed file, are readable. However from the end of the just recovered file to the end of the PCMCIA card will now be full. In order to regain the space on the card, copy the desired portion of the last take (which presently uses up the remainder of the card) to another card, by means of the editor, and simply delete the take from the first card.

When F2 is pressed the following display will normally be seen:



ESC (F4) is Escape as on a PC which will put the display back to the main screen.

5.0 REAL TIME CLOCK (R.T.C.)

The RTC of the ARES-C is an internal clock with both time and date and will stay correct as long as there is sufficient power from the batteries. This value will be kept in memory for several minutes to allow changing flat batteries. Pressing F3 (RTC) from the main display will display the following:

```

DATE & TIME
          13:59:03
          12:07:98
TIME      DATE  ESC
    
```

Pressing F1 (TIME) will change the display of the time portion to HH:MM:SS and the left-most digit (hours tens) will be flashing. Enter the time in the 24 hr format using the numerical keys. Once the seconds unit has been entered the new time will be displayed (a few seconds later) and the RTC will have been set. Pressing F3 (DATE) will give the opportunity to set the date portion of the RTC. The date must be entered in the DD:MM:YY format. Once the setting is complete press F4 (ESC) which will return to the

previous screen, continue pressing ESC until the main menu is once again displayed.

Correct setting of the internal RTC is important as it is the RTC which is used to indicate the time and date of the recordings made. This information is stored in the Directory of the card.

5.0 bis VERSIONS

```

DECK      :      X.XX
MOTHER BOARD :      X.XX
DSP TYPE   :      2
ISDN      :      X.XX
          ESC
    
```

Pressing F5 (VER) shows the eeprom versions as well as the type of DSP inside the ARES-C.

6.0 DIRECTORIES (DIR)

In order to explain this section, it is assumed that the PCMCIA card has some data (takes) recorded in it. Turn the editor ON by pressing F4 as explained above then press F1 (DIR) and the following display will be shown. Naturally the numbers below are just for this example and will be different depending on the information in the card.

```

CARD IN USE
          PRESENT TAKE N°
A:      1      0:04
000     0.00   13/06/95
> 001     0.14   13/06/95
002     0.07   14/06/95
003     0.40   14/06/95
END     xxx
EDT     DEL   A-B   ESC   PLY
    
```

TIME FROM START OF TAKE N° 1

DATE CARD WAS FORMATTED

DATE THE RECORDING WAS MADE

TAKE DURATION

REMAINING TIME ON CARD

TAKE LISTING

PRESENT POSITION

From the left display it can be seen that card A has been selected. It contains 3 recorded takes plus the format take 000. The machine is presently at a position of 4 seconds after the start of take 1 that is a take lasting for 14 seconds recorded on 13 June 1995. If the right arrow key is pressed the time of the particular recording is shown. Pressing the right arrow again will indicate the compression format in which each take was recorded by the entire display scrolling to the right. If the right arrow key is pressed again, the title area is shown. At the bottom of the display there are five possible choices on the function keys. F1 (EDT) will switch to the edit mode,

F2 (DEL) will allow a take to be deleted, F3 will toggle between the two PCMCIA cards, F4 (ESC) will return to the previous screen and F5 (PLY) will put the machine into playback from the exact point where it is at this moment. Each of these features is explained in more detail below.

By means of the UP and DOWN arrow keys the cursor can be moved to the desired take. If the SHIFT key and one of these arrow keys are pressed at the same time then the cursor will jump to the top or bottom of the list accordingly. Moving the cursor to a specific take number is possible by simply entering the three digit take number using the numerical keypad (thus take #3 should be entered as 003). At the bottom of the directory listing the total time remaining on the card is indicated, according to the compression mode presently selected.

NOTE: The directory of the ARES-C can store up to 408 different "Takes". If the user tries to record take #409 it will be recorded on the second card if it is installed and providing there is sufficient space. If this is not the case then the machine will not switch into record. If an EDIT is attempted on a card with 408 takes recorded the display will indicate "CARD FULL".

7.0 PLAYING BACK A SELECTED TAKE

Move the cursor to the desired take using the arrow keys, and press F5 (PLY). The ARES-C will immediately start to play this take from the beginning as long as the compression mode presently selected is the same as that of the take, the green READ led corresponding to the card in use will light and the display will be as shown below:

A:	1	0:04
000	0.00	13/06/95
▶ 001	0.14	13/06/95
002	0.07	14/06/95
003	0.40	14/06/95
<<	>>	< > STP

The five function keys now have the following operations:

- F1 Rapid rewind at 40 times nominal speed while key is held down
- F2 Fast forward at 40 times nominal speed while key is held down
- F3 Rewind at 4 times nominal speed while key is held down
- F4 Forward at 4 times nominal speed while key is held down
- F5 PLAY / STOP toggle switch

NOTE: Indicated in the right-most column when the right arrow key is pressed twice is the compression mode that was used when the original recording was made. If during playback the machine is set to G722 then it will only play G722 takes and will automatically jump over all other takes.

8.0 DELETING TAKES

From the directory display F2 (DEL) corresponds to the DELETE function. When this key is pressed it will delete the selected take **AND ALL TAKES THAT FOLLOW IT**.

Move the cursor using the up and down arrow keys, or enter the desired take number using the numeric key pad (in this example take 1 was chosen), and then press F2. The following will be displayed:

```

SLOT A
DELETE TAKE 001
ONWARDS ?

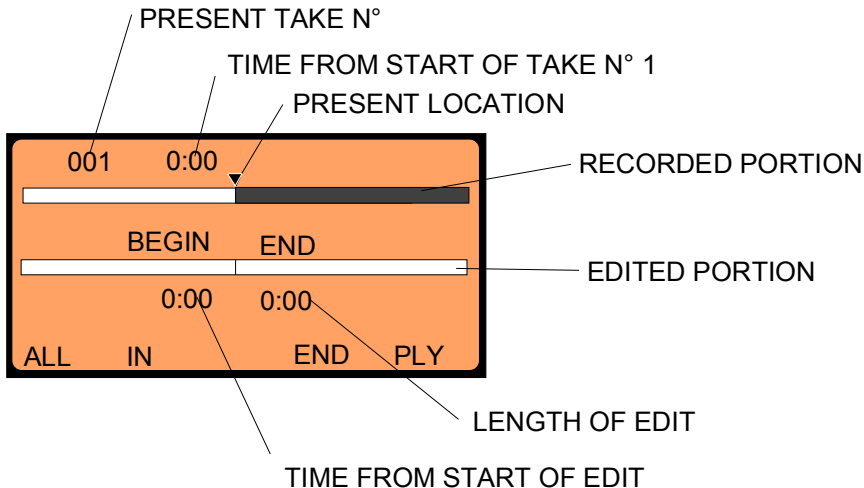
YES          ESC
  
```

If F1 is pressed then Take 1 and all the takes following it will be erased from the card. If F4 (ESC) is pressed then the erase will not occur and the display will return to the directory display.

NOTE: Erasing a PCMCIA card is not like a floppy disk. That is to say if an erase or format is made then the card is overwritten and it is impossible to make any form of "unerase". It should be considered in the same way as a magnetic tape in this respect.

9.0 OPERATION OF THE AUDIO EDITOR

From the DIRECTORY display, press F1 (EDT) and the display will switch to the following:



When this screen is displayed then the function keys have the following operations:

- F1 ALL is to select the entire take for editing
- F2 Toggle to mark IN / OUT points (CUE 1 / CUE 2)
- F3 Not used
- F4 End editing session
- F5 PLAY / STOP toggle switch for the selected tape (source or target).

9.1 PRINCIPLES OF EDITING

When using the editor of the ARES-C there are two principle operations that can be performed. One is the editing together of various different sections of different takes that have previously been recorded, the other is the insertion of a selected section from one take into another take. Both of these methods will be explained in detail. A final point, which will be covered, will be the modification of edit points and their elimination (UNDO). In this section the graphic displays will be referred to as "tapes" the upper tape being the "source" tape and the lower representation being the "target" tape.

9.2 HOW THE EDITOR OPERATES

It is important to understand how the editor operates before trying to make edits. The information that is recorded in the PCMCIA card is recorded as tiny blocks of data, each of which is clearly labelled. During normal playback these sections are read from the card in their numerical order and then played back. Editing made on data in the PCMCIA card is known as VIRTUAL editing. This means that unlike cutting and splicing 1/4" audio tape the original material remains entirely intact. With traditional editing, the useless portions of the recording are discarded and only the useful parts are kept, with virtual editing the original is untouched.

Editing is made by the creation of an Edit Decision List (EDL) which contains all the desired IN and OUT points. When the EDL is played, the list of locations of the various sections of audio required is fed to the microprocessor, which then retrieves each section in turn from the original data in the PCMCIA card.

Virtual editing has great advantages over conventional editing in that as the source material remains unchanged, many different EDL's can be created with the same source material until the desired result is reached. The ARES-C's editor will store each set of edit instructions (EDL) like a take that will be numbered consecutively after the last recorded take. The only differences are that it is marked as an EDL in the directory listing with a tiny pair of scissors next to the take number in the directory listing, and is displayed

with "dots" on the tape.

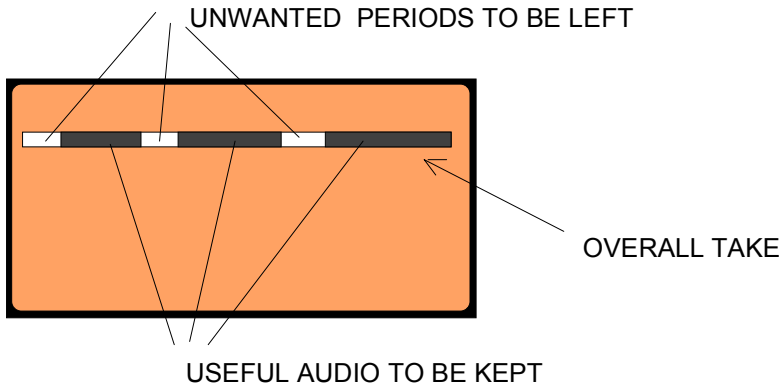
It should be noted that an EDL takes very little space in the PCMCIA card and even if a card is full (with audio) it is still possible to store several EDLs in the remaining space on the card. It should also be noted that it is not possible to select portions of audio from the second card for an EDL. If information is required from the second card it must be copied to the other card before editing is started. Finally, it is not possible to combine pieces of audio which were recorded using a different method of compression.

9.3 STEP BY STEP EDITING ON THE ARES-C

A step by step guide through some simple examples of features possible using the editor will be enough for the user to fully understand the facilities of the entire editor. The explanation will be broken down into several different simple examples.

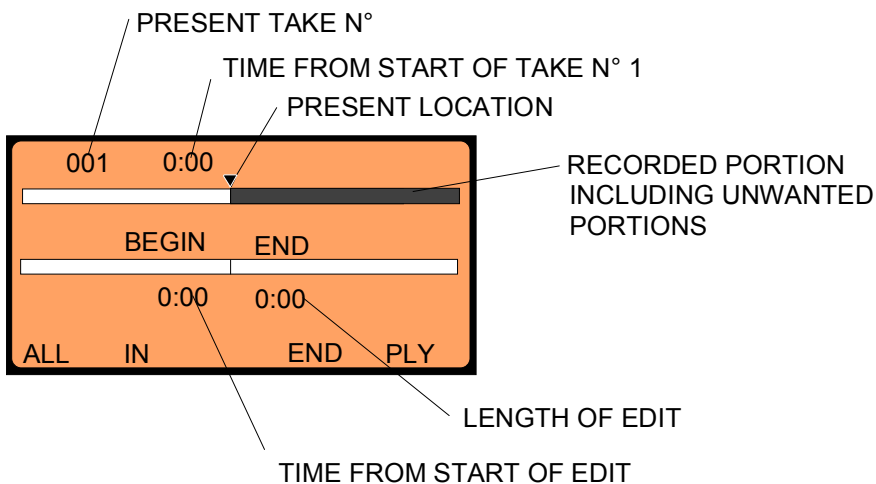
9.3.1 EXAMPLE 1

In this first example, imagine that there is only 1 take on the card which was recorded over several minutes, and there were long periods of unwanted audio between various pieces of useful audio. Our initial goal is to select the useful portions and create an EDL which is a fluent reproduction of the recording ready for transmission. There are two methods of performing this operation each of which will be covered. A graphic indication of such a recording would be as follows:



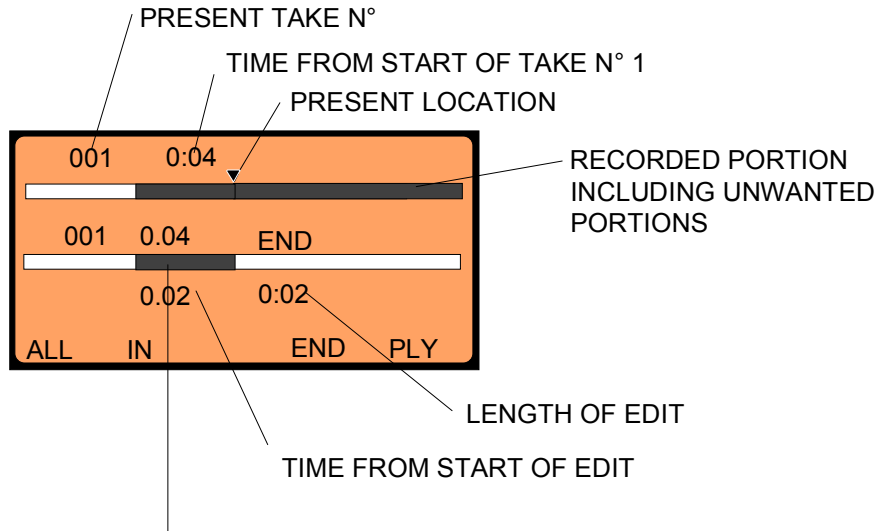
This is not an actual display but is for this example.

Turn on the editor of the ARES-C and from the directory screen press EDT to activate the editor. The screen would actually then be:



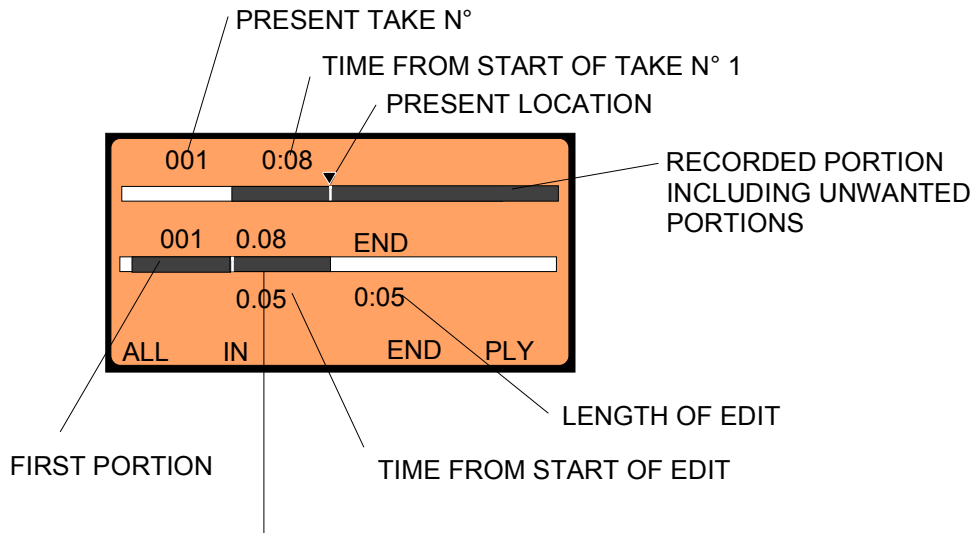
If F5 (PLY) is pressed then the entire take will be played from the beginning of the take. Equally if the JOG wheel is turned the take will advance as the wheel is turned. The audio can be heard through

either the headphones or the internal loudspeaker. Continue to turn the JOG wheel until the first useful piece of audio is located. Once the start of the useful audio is located press F2 (IN) which will mark this point as the IN point for the first edit (once pressed F2 will become the OUT point marker key). Continue to turn the JOG wheel until the end of the first piece of useful audio is found and then press F2 (OUT) which will mark this as the edit out point for the first edit. The selected portion of the source tape is copied "virtually" onto the target tape. Pressing the shift key while turning the JOG wheel will jog at high speed in either direction. The screen should then display:



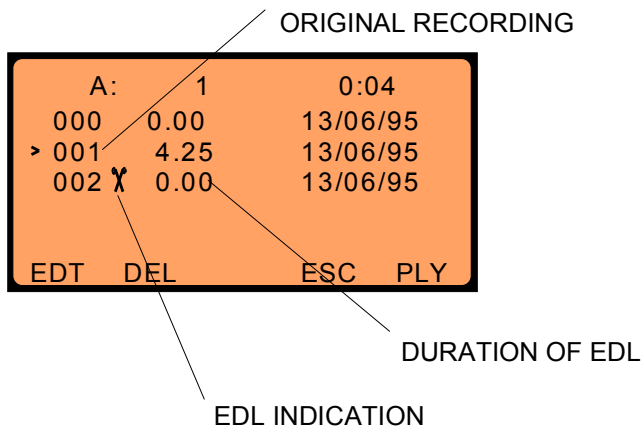
NEWLY SELECTED PORTION BETWEEN THE EDIT IN AND OUT POINTS

Now continue to turn the JOG wheel until the unwanted portion is over, and then press F2 again which will mark the second IN point. Continue as before until all the useful audio has passed, then press F2 (OUT) and this section will be added after the initial section:



NEWLY SELECTED PORTION BETWEEN THE EDIT IN AND OUT POINTS

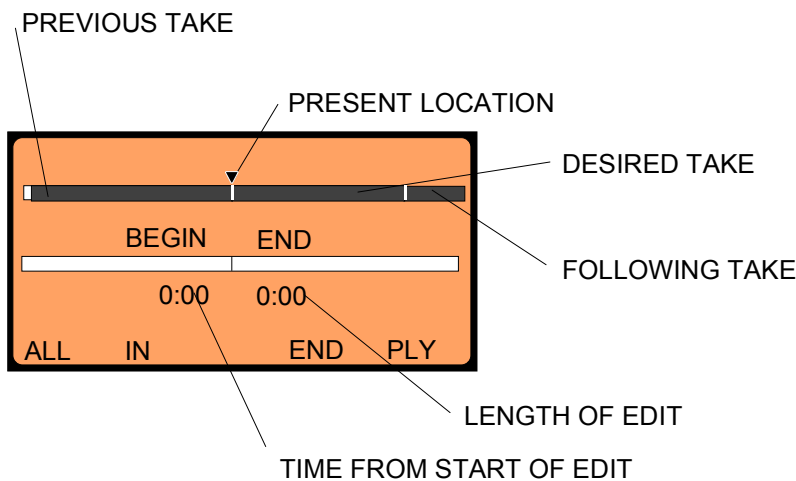
Proceed in this manner until all the wanted portions of audio have been extracted. Press the DOWN key and the cursor will jump to the lower (newly created) portion. It is then possible to play the edited section. Once this editing is completed press F4 (END) and the screen will ask if this is to be saved as take N° 2 and t Proceed in this manner until all the wanted portions of audio have been extracted. Press the DOWN arrow hen press F1 for YES. Pressing escape will then return to the previous menu and the newly created EDL can be seen in the directory. The display will be:



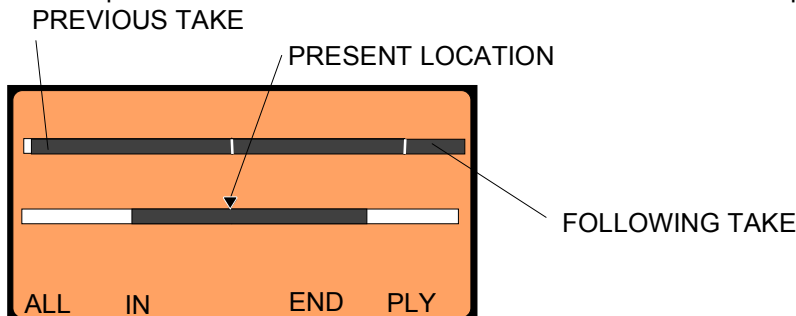
9.3.2 EXAMPLE 2

In this example let us assume that a complete take is to be selected and then a portion of audio from the following take is to be edited into the middle of the first take.

Turn on the internal editor and move the cursor to the desired take in the directory listing and then press F1 (EDT). Alternatively the editor can be turned on first, and the desired take located by pressing the right arrow to skip forwards, or the jog wheel to move along the simulated tape at the top of the display to the desired point. Once it has been located, the display will look like this:

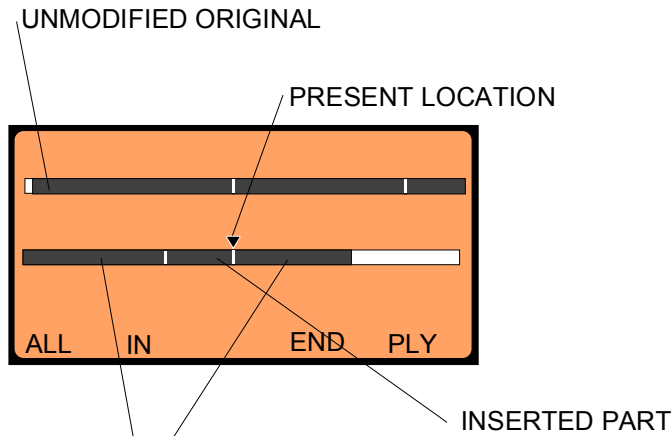


Here, the cursor is at the beginning of the actual complete take to be copied. The preceding and following takes are also displayed. Press F1 "ALL" and the entire take will be copied to the lower "tape". Press the down arrow key and the cursor will move to the lower tape. Move the JOG wheel until the point where the other information is to be inserted. The display will show:



Pressing the UP arrow will move from the lower target tape to the upper source tape again, using the jog wheel locate the start of the section of the following take which is to be edited into the take previously selected. Once the edit IN point has been located, press F2 (IN) then, using the JOG wheel locate the end or edit OUT point and press F2 (OUT) again. Pressing the down arrow will move the cursor to the target tape. This section will then be inserted into the previously selected take

and the screen will show:



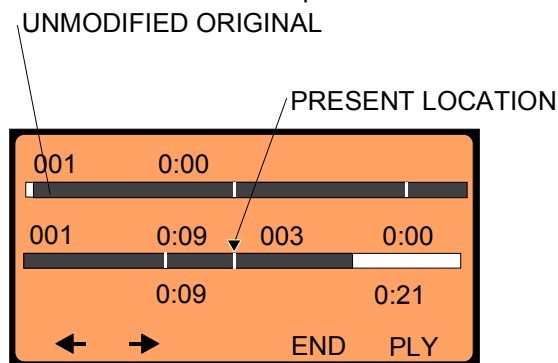
START AND END OF ORIGINAL SECTION

It is now possible to skip to the beginning of the lower "tape" using the left arrow key, then to either PLAY the entire section using F5 (PLY) or to move slowly along the section using the JOG wheel. Once happy with the results, press F4 (END) and the display will once again ask if this is to be saved as Take N° X, if YES is pressed then it will be added to the EDL in the directory listing.

9.3.3 EXAMPLE 3

In this example, the goal is to take a previously edited section in which one (or more) of the edits made is not exactly at the correct place. This will explain how to move an edit point or even to delete an unwanted section of audio.

Switch on the internal editor and go to the directory screen. Using the down arrow move the cursor to the EDL to be modified and press F1 (EDT). Immediately start of the edited section (indicated by a dotted tape) will be shown. Press ALL to select the edited section and it will immediately appear with all its edit points on the lower tape. The cursor can be moved down to this tape using the down arrow and it will be at the end of the edit. Move to the edit point that is to be moved using either the JOG wheel, the arrow keys or F5 (PLY) key. Position the cursor exactly on the edit point. If the cursor is to the left (before the edit point) then only the take number and position in the left take will appear, if the cursor is just to the right of the edit point then only the information relevant to the right portion will be displayed. When the cursor is perfectly on the edit point both sets of information will be shown. The screen below shows this position:



To modify the edit point (marked "Present location" above) it is very simple to slide the right or left portion of the tape. Pressing F1 means that the left piece of tape will be moved and pressing F2 will move the right portion. When F1 or F2 are pressed, an additional timer will appear in the bottom center of the screen and will increase or decrease as the JOG wheel is turned to indicate how much time is being added / removed from the take.

Simply press the corresponding key and then while keeping it depressed gently turn the JOG wheel to the desired new place. Once the correct edit point is located release the F1 / F2 key. When the modifications are finished press F4 (END) and the screen will once again ask if the EDL is to be saved as take N° X. Press YES / NO accordingly and then ESC to return to the main menu.

Finally, the removal of an edit point is possible by moving the cursor to the point (the edit point is located when both sets of duration numbers appear), and pressing the F3 (DEL) key, the edit point will then disappear. If the DEL function is not available on the display then this edit point is not one that can be removed by simply pressing DEL.

Removal of a complete section between two markers, simply place the cursor anywhere within the take (between the two markers in question) and press DEL.

CONCLUSION

Three examples have been covered here and once these principle manoeuvres are understood then almost all the possibilities of the editor of the ARES-C can be fully understood. Naturally all other types of edits, be it removing a sound, or moving a take to another position, moving of edit points etc. can all be performed using the above mentioned principles. The important points that have been covered here are:

- Simple Cut and Paste from the original to the edited version
- Copy of a complete take into the EDL
- Insert a portion of audio into another take
- Moving of an edit point to the left or right
- Removal of an edit point
- Removal of a take

9.4 EDIT INDICATION

In the directory listing on the main display and EDL is indicated by a small pair of scissors next to the take number. When the editor is switched on the graphic representation of the tape is shown. All source material (original recordings) are indicated as `////////` and `\\\\\\\\\\\\` alternatively. All EDL's are graphically indicated as either `/././././` or `\\.\.\.\.`

9.5 A-B CARD SELECTION

When the editor is switched on and the directory listing is displayed, the function key F3 is allocated to the A-B selection. Each time it is pressed then card A or B will be selected alternatively.

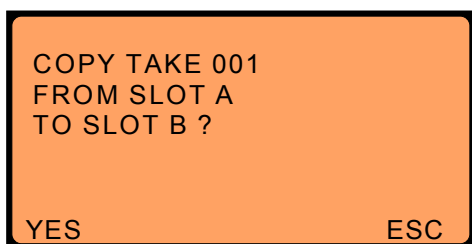
10.0 THE COPY FUNCTION

The copy function of the editor of the ARES-C is used to copy takes from one card to another. The copy function is activated by pressing the "SHIFT" key #4 on the top deck plate while in the directory display. When the directory is displayed if shift is pressed then the functions under the function keys disappear and only F1 "CPY" is available.

A:			
	1	0:04	
000	0.00	13/06/95	
▶ 001	0.14	13/06/95	
002	0.07	14/06/95	
003	0.40	14/06/95	
CPY			

Will appear if "SHIFT" is pressed

While still holding the shift key down press F1. In the above example the following screen will be displayed:



If F1 “yes” is pressed then the desired take will be copied to card B. The take to be copied is chosen by moving the cursor using the up and down arrows to the desired take before pressing the shift key. If a previously edited section is copied then the corresponding source material is also copied and the whole edit is stored on the other card in the form of a new take. If the copy function is attempted when there is no second card in the machine then the error message "SLOT NOT READY" will be displayed. CARD FULL will be displayed if the user tries to copy audio from one card to the other and there is not enough free space on the receiving card to accept the entire take selected.

11.0 EDITOR ABBREVIATIONS

The following list shows all the possible commands that may appear next to the five function keys when using the internal editor on the top deck plate. A brief description is also indicated.

A-B	Card selection toggle. This will only operate if the machine has two PCMCIA cards installed. Card "A" refers to the upper slot and "B" the lower slot.
ALL	ALL selects the whole take for editing. This function is to be used if a previous edit is to be reworked.
CHK	Check format of the selected card. This should be used if an error message appears on the display indicating that there is an error with the data recorded in the PCMCIA card.
CLR	Clear incorrectly entered data when entering a telephone number.
CONT	Continue is used to pass to the next screen when using the ISDN option.
CPY	Copy is used to copy takes from one card to another. This function will only appear when the SHIFT key on the upper deck plate is pressed.
DEL	Delete can be used to delete either an edit point if the editor is on, an entire inserted section in an edit, or to remove all takes from the present point onwards while in the directory listing.
DIR	Directory listing will show all the takes recorded on the selected card as well as the duration, start time, date and compression mode.
EDT	Edit is used to edit the selected take when in the directory listing.
END	END editing session
ESC	Escape to last screen
FOR	Format card
IN	Edit IN point mark
ISDN	ISDN is used to select the digital phone output connector. This message will only appear on machines fitted with the ISDN option
LIN	Line output is used to select the Line Output as the destination when transmitting data via the telephone output.
MIS	Miscellaneous gives access to various other setting within the machine such as RTC and Formatting of the cards.
ON / OFF	Used to turn the editor ON or OFF.
OUT	Edit OUT point mark
PLY	Play
RCL	Recall a telephone number stored in the memory.
REC	Record is used to record information coming to the ARES-C from either the analog telephone line or the ISDN input.
RTC	Real Time Clock gives access to the setting of both time and date of the internal real time clock.
SEL	Select a take to be transmitted.
TEL	Telephone output selection.
TRM	Transmit is used to send takes either to the analog outputs, the PSTN telephone output or to the ISDN circuit for transmission onto the digital network.

SET	Settings. Give access to modify or to verify the configuration of the machine.
STO	Store a telephone number in one of the 10 internal memories.
STP	Stop while playing back a take.
< OR >	4 X nominal speed in either direction. In this mode audio will still be available on the headphones and loudspeaker.
<< OR >>	40 X nominal speed in either direction. In this mode audio is NOT available.
<- OR ->	Edit selection is used to select the portion of audio either side of the edit point for "Stretching" or "Shrinking". These keys are also used to mark new edit points in a take. They can be used "on the fly"
VER	Version. Shows the software level as well as the type of DSP used inside the machine.

12.0 ERROR MESSAGES

12.1 Deck plate display

When the machine is turned on it is possible that an error message will be displayed on the main display. The possible messages and their explanations are:

UNFORMATTED CARD	A new card has been installed on the machine and it must be formatted before any recording can be made.
INVALID CARD	A card of an unknown type has been installed into one of the slots. Replace the card with a known card.
UNKNOWN FORMAT	Information has been detected in the card but this data does not correspond to the ARES-C's format (such as a PC's format). Reformatting the card is the only solution.
FORMAT CORRUPTED	This indicates that the format is the ARES-C's format but there is a problem. Turn on the editor and use the CHK (Card Check) function described below.

NOTE: If at any time when using the ARES-C the message SAVE ERROR appears on the display when trying to write information to the card, check that the WRITE protect switch on the end of the card has not been placed in the protected position. If it has then simply use a small pointed instrument to move the switch without turning off the machine or removing the card, then try to make the save of the information again. If this was the problem then the display will indicate SAVE COMPLETE.

Other error messages will appear on the main display when certain actions are performed. These messages are totally self explanatory. For example if a copy is attempted from card A to card B but card B is already full, then the display will indicate CARD FULL when the copy key is pressed. The messages that may appear are as follows:

WRONG COMPRESSION
 CARD WRITE PROTECT
 CARD FULL
 DIRECTORY FULL
 SLOT NOT READY

12.2 Front panel display

SET.LOST This will appear if the machine has been left without power (internal batteries) for more than a few hours, or if the Eproms within the machine have been changed for a newer version. To remove this message from the display press SHIFT + EXE on the front panel.

NOTE: After a SET LOST message the machines internal settings will resort to the factory default settings and this will include the Real Time Clock, which should be reset.

LOW BAT.	Will be displayed when the internal batteries become too low.
Rec. on B	This will be displayed if card A has been selected, and it is already full when the machine is put into the RECORD mode. It is a warning that the machine is recording on card B, not A.
Rec on A	As above but for when card B is full.
A Full	Will be displayed when there is only 1 minute of recording time left on card A.
B Full	Will be displayed when there is only 1 minute of recording time left on card
Card err.	Appears if a card error is detected during record or play

12.3 BEEP MESSAGES (Headphones and / or speaker)

1 Beep	Execute of a function in the menu mode (function accepted) Error message has been displayed on the front panel display Only 1 minute of recording time remains on the card presently being used
2 Beeps	Low Batteries has been detected Function execute refused in the menu mode
3 Beeps	The DSP is not responding anymore in the record mode. The red led's will turn off immediately if this error occurs.
4 Beeps	Internal settings have been lost (SET LOST on front display)

NOTE: It is possible that two or more of the above occur at the same time and therefore more beeps will be heard. For example, if the recording switches over to a second card which has only 1 minute left of recording time and at the same moment the low bat indication is given. In this case multiple beeps will be heard.

12.4 FLAGS F1 to F8

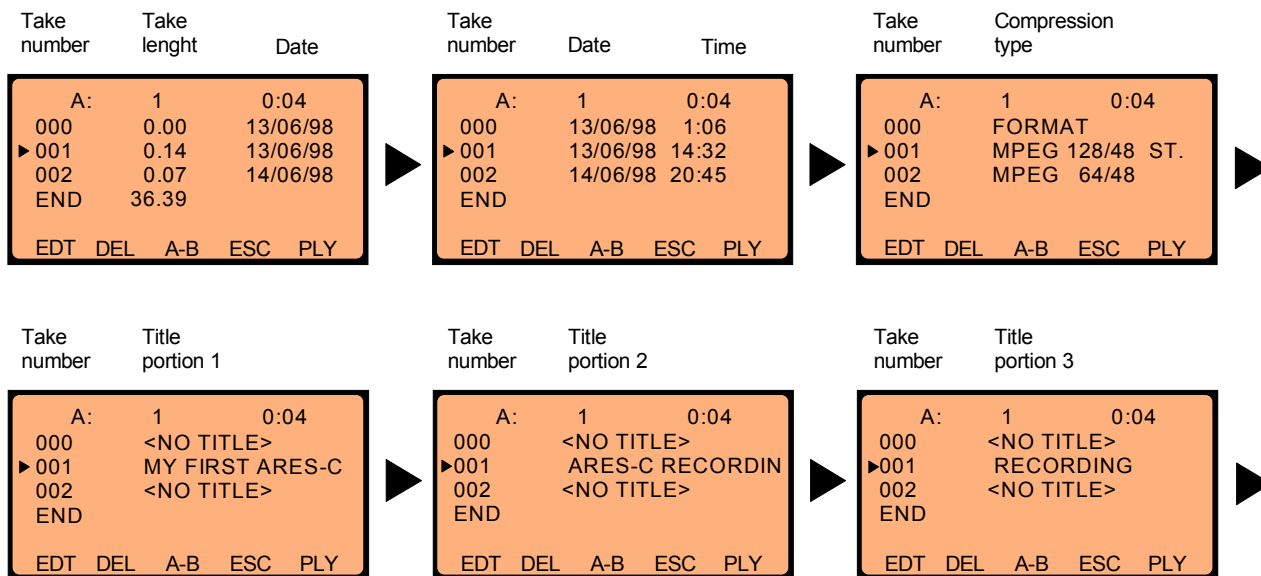
F5	Will light up at the moment that the battery voltage per cell becomes 1V.
-----------	---

13.0 TITLING

Set the ARES-C main front selector to "EDIT / STD-BY" mode and press the F1 key "DIR".

13.1 HORIZONTAL DIRECTORY SCROLLING

Pressing the right or left arrow key allows scrolling the directory information horizontally.



Once "DIR" is selected, followed by pushing only the right arrow key several times, the above figure will be displayed step by step. If the left arrow key is pressed, the order of displaying will be reversed.

13.2 VERTICAL DIRECTORY SCROLLING

This can be done by using the up or down arrow keys to move step by step inside the directory. If the up or down arrow keys are combined with the shift key, the display will jump to the beginning or the end of the directory.

It's also possible to move inside the directory by introducing the take number on the numeric keypad. Example: introducing "0 2 3" on the keypad will display the take number 23.

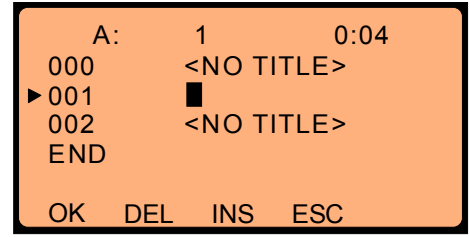
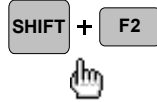
13.3 INTRODUCING A TITLE

A new title for the format file 000 or any other audio take can only be entered if the file has "NO TITLE". On flash cards "Series 2, 2+ and Strata", once the title entered, it can not be modified anymore. In the case that a title needs to be modified, the only possibility is first, to make an edit of the file and save it, secondly enter a file name for the edit.

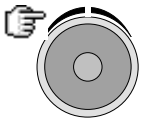
On ATA flash cards, the title of any take can be modified.

Using the up or down arrow keys select the file that needs a title. Verify that in the title portion 1, 2 or 3 <NO TITLE> appears.

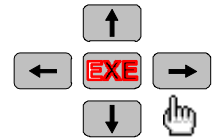
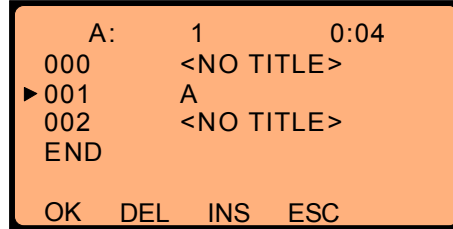
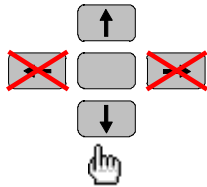
Press the shift button together with F2 "TITL". The following screens will be displayed:



A little square starts blinking at the position of the first character.



OR



Turning the jog-wheel or pressing the up or down arrow keys scrolls through:

A B C D E F G H I J K L M N O P Q R S T U V W X Y Z 1 2 3 4 5 6 7 8 9 0 ! # \$ % & ' < > + - = , "Blank"

The numerical keypad can also be used to introduce numbers.

Pushing the right arrow key once or the "EXE" key, jumps to the next character position.

To insert a character in a chain, move the blinking square to the position for insert and press F3 "INS".

To delete a character in a chain, move the blinking square to the position to delete and press F2 "DEL".

Once the title is introduced, press F1 "OK". This will add the title to the associated file. No more title changes are possible once the "OK" has been pressed.

Important: A title can have maximum 31 characters (a "SPACE" is counted as a character).
 Titles once introduced and confirmed by "OK" can not be modified anymore.
 If an EDL of a take is made, a new title can be introduced.
 Copying a take from one card to the other will also copy the title.

CHAPTER IV TRANSMISSION

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General

The ARES-C is equipped with various different outputs allowing it to be used in many different circumstances. These outputs include a standard analog output, a 600 Ω telephone output, an AES bus and an optional ISDN connection or optional time code connection. Each of these is covered in more detail below. This chapter describes the ISDN transmission. For Time Code, refer to chapter IV addendum.

1.0 OUTPUTS

IMPORTANT: If MPEG ST. is selected, the DTMF signal is available only on the right channel.

1.1 ANALOG OUTPUTS

The analog outputs of the ARES-C are transformerless AC coupled and are located on the right-hand side panel on two male 3 pole XLR connectors. The standard output level corresponding to 0 dB on the meter is 1.55 V. This output level can be modified using the potentiometer "Aux IN and Line OUT" on the front panel providing it has been selected in the menu mode (see chapter II of this manual).

The DTMF generator is enabled once the "LIN" menu is selected.

The * tone is generated when "SHIFT" & "1" is pressed, the # tone is generated when "SHIFT" & "2" is pressed.

1.2 AES OUTPUT

The 3 pole male XLR AES output connector is a digital output corresponding to the format of the AES bus used throughout the professional audio industry. This output is only available if the compression mode used was MPEG-1 layer II and the sampling frequency during the recording was either 32 kHz or 48 kHz otherwise it is disabled. The resolution is 16 bits. This connection allows direct digital link to any other digital equipment equipped with an AES interface.

1.3 TELEPHONE OUTPUT

The telephone output is on three banana connectors on the right-hand side of the machine. This output has a transformer and its level is selectable by means of the internal menu mode. The possible output levels representing 0 dB are 4.4V and 1.55V no load (menu selectable) with a frequency response from 300 Hz to 5 kHz. When connected to the telephone line, the return signal is available on headphones or the internal loudspeaker (according to the selection).

The DTMF generator is enabled once the "TEL" menu is selected.

The * tone is generated when "SHIFT" & "1" is pressed, the # tone is generated when "SHIFT" & "2" is pressed.

2.0 ISDN

The ISDN connection is available if the ISDN option is installed in the machine. The connection is made on the right-hand side of the machine using a RJ 45 type connector. The ISDN is a digital network designed as the successor to the existing Public Switched Telephone Network that offers much greater flexibility for data transmission. It allows a greater bandwidth to be transmitted (7kHz with G722 compression as opposed to 3.1 kHz with the Public Switched Telephone Network). It allows data transmission at 64 kbits/sec. There are a wide range of applications that it can be used for, in the case of the ARES-C it permits high quality audio, text or even graphics files to be sent between the ARES-C and the base station (not presently possible). A standard ISDN connection that can be found in the field (at a sports hall or conference centre for example) will generally be a **2B+D** connection. This means it has two 64 kbits/s channels for voice/data known as the "B" channels and one "D" channel used for signalling. The ARES-C will connect directly to such a line. Generally one B channel is used for mono transmission and two "B" channels would be used for transmitting stereo information. The "D" channel is used for data such as the number being called, the taxation of the line etc.

The DTMF generator is enabled at the moment the ISDN connection is on line.

The * tone is generated when "SHIFT" & "1" is pressed, the # tone is generated when "SHIFT" & "2" is pressed.

The basic principles of an ISDN system are explained in more detail in the appendice to this manual.

3.0 SENDING INFORMATION FROM THE ARES-C TO THE OUTSIDE WORLD

Once the recordings have been made and in turn edited on the ARES-C it is possible to send these files from either of the PCMCIA cards to the outside world. In order to do this, turn on the internal editor and on the first screen that appears press F2 "TRM" (transmit) and the following screen will be displayed:

```

SELECT TRANSMIT MEANS
F1: LINE OUT
F2: ANALOG PHONE
F3: DIGITAL PHONE
F5: ISDN SPEECH MODE
  LIN  TEL  ISDN  ESC  SPM
  
```

3.1 TRANSMISSION TO THE ANALOG OUTPUTS OF THE ARES-C

If F1 is pressed then the display will show

```

MUTED  EE  MIX-OFF

MUTE DIR          ESC
  
```



```

      A: 001      000
000   0.00     10/12/97
▶ 001   2.45     11/12/97
002   1.30     11/12/97
000   0.00     10/12/97
END   32.46
SEL           A-B  ESC  PLY
  
```

If F1 "MUTE" is pressed then the display will show "MUTED EE MIX-OFF" (the Mic. inputs as well as the line inputs are muted). F4 "ESC" allows the user to return to the previous screen. "EE" appears if the EE/AUTO/TAPE switch on the front panel is in the EE position. "MIX-OFF" appears when during the play mode the line inputs as well as the microphone inputs are muted. Using the up and down arrow keys permit to switch on or off the mix mode. The DIR position F2 will display the directory of the presently selected card and then the F1 position will be marked "SEL" (select). The letter A or B in the top left corner of the display indicates the presently selected card.

If the cursor is moved to the desired take number and the F1 key is pressed the display will change to:

PRESENT CARD SELECTED
ELAPSED PLAY TIME FROM START OF TAKE N° 1
PRESENT LOCATION

```

A: 001 0:00 R 2.45
  ───────────▶
001 0:00
MIX-OFF

MUTE DIR          ESC  PLY
  
```

REMAINING PLAY TIME OF THE TAKE

F5

The remaining time will count down to zero

```

A: 001 0:11 R 2.34
  ───────────▶
001 0:11
MIX-OFF

MUTE           STP
  
```

as pressing the PLY key F5 is transmitting the data. At any time the STP key F5 can be pressed during the transmission to momentarily interrupt the transmission. When the end of the segment to be transmitted is reached then the screen will be as above except it will be at the end of the recorded take with remaining time 0:00, and the microphone or Aux. input (depending on selection) becomes active again. In the stop mode, the left and right arrow keys permit to play backwards or forwards at 4x nominal speed. The centre key (EXE) permits to jump to the beginning of the take.

3.2 TRANSMISSION DOWN AN ANALOG TELEPHONE LINE

The method is the same as used for the analog outputs except F2 is pressed to select ANALOG PHONE as the destination for the data.

It is now possible to use the ARES-C to dial the required telephone number (DTMF only) when working on an analog telephone line. This is simply done by connecting the banana outputs to the analog telephone line. It is important to know that, once the bananas are connected to the line, that the line is activated (no relay inside the ARES-C). The telephone number can be dialled only straight from the numerical keyboard.

The * DTMF tone is generated using the "SHIFT key together with the "1" key.

The # DTMF tone is generated using the "SHIFT key together with the "2" key.

Once the ARES-C is connected onto the line, then the microphone and the headphones are operational, and the return signal of the line will automatically be heard in the headphones, depending of the position of the TAPE / AUTO / EE selector. **This selector must be in the "AUTO" position.** It is for this reason there is the possibility to MUTE the input so that the microphone can be turned off if desired. In any event the microphone is muted automatically whenever a file or take is being transmitted down the line. **Once the communication is finished then the ARES must be disconnected from the line in order to cancel the communication. If this is not done then the line will be kept open even if the ARES-C is switched off.**

3.3 TRANSMISSION / CONNECTION TO AN ISDN NETWORK WITH DSP II

An introduction to the basic operation of an ISDN network is covered in more detail in the appendice to this manual. **Before an ISDN transmission is made, be sure to check that the ISDN sub-menu in the SET menu is set to the correct positions.** If a take (or edit) is to be transmitted over the ISDN network, turn on the editor and press F2 "TRM" followed by F3 "ISDN" the following displays can appear:

The left display will appear if the selected compression is G.711, G.722 or MPEG **uplink-downlink**.

```
SELECT TRANSMIT MEANS
F1: LINE OUT
F2: ANALOG PHONE
F3: DIGITAL PHONE
F5: ISDN SPEECH MODE
LIN TEL ISDN ESC SPM
```



```
TAKE PRESELECTION
F1: PRESELECT A TAKE
F2: NO PRESELECTION
DIR CONT ESC
```

ATTENTION: If F5 "ISDN SPEECH MODE" was pressed instead of F3, the same procedure can be followed as for F3 "DIGITAL PHONE" but the ISDN connection will be made in G.711 A-law or μ -law. This means that without changing the settings of the machine an ISDN call to a standard PSTN telephone can be made.

If MPEG-G722 or MPEG-G711-A was selected, the following display will appear.

```
TAKE PRESELECTION
F1: PRESELECT A TAKE
F2: NO PRESELECTION
F3: UPLOADING
DIR CONT UP ESC
```

An ISDN communication in MPEG mono or stereo at 128kb/s or 192kb/s is not possible actually. However, once a 64kb/s ISDN connection is made, the ARES-C will convert MPEG mono or stereo files recorded at any bitrate and with the same sampling frequency versus the "SET" menu, to a second generation 64kb/s mono file for the transmission.

If F1 "DIR" is pressed then the directory listing of the current card is displayed and the user can then pre-select a take using the arrow keys. Pressing F4 "ESC" will return to the main menu. Pressing F2 "CONT" (continue) means that the user wishes to make an ISDN connection without pre-selecting the take to be transmitted. This would be the procedure if for example the user wants to receive something via ISDN rather than send a message.

If F3 "UP" is pressed (if MPEG mono 64kb/s is selected), the ARES-C becomes ready to receive in the MPEG mode and to transmit in the G.711 or G.722 mode. If only F1 or F2 would be pressed, the ARES-C will be in a status for transmitting MPEG and receiving G.711 or G.722. The selection of G.711 or G.722 combined with the MPEG compression is made in the SET, ISDN sub-menu.

IMPORTANT

In the case of MPEG transmission at 64kb/s, F3 "UPLOADING" means that the ARES-C is set to emit a G.711 or a G.722 algorithm and to receive a high quality sound in the MPEG algorithm. If F3 "UPLOADING" is not pressed, the ARES-C is set to emit in a high quality MPEG algorithm and to receive in a G.711 or a G.722 algorithm (see ISDN menu in chapter II).

If from the previous display, F2 is pressed then the following screen is displayed:

```
SELECT DIAL MODE
F1: INCOMING CALL
F2: AUTO ANSWER
F3: DIAL A NUMBER

INC  AUTO  CALL  ESC
```

Again, F4 "ESC" will return to the principle menu. Pressing F1 will allow the operator to put the machine into standby mode in order to await an incoming call on the ISDN line. When the call comes in, if the speaker is not switched OFF, the ARES-C starts beeping until F2 "ANS" is pressed. Pressing F3 "CALL" will allow the operator to dial a number from the ARES-C.

F2 "AUTO ANSWER" can be used in two different ways depending on the previous menu:

If in the previous menu no pre-selection was made, the ARES-C will be in an automatic receiving status. This means that every time the ARES-C is called, it will automatically go in record until the end of the communication. At every call, a new take number is automatically created.

If in the previous menu a take of the directory was selected, after 5 seconds it will automatically go into play mode from the beginning of the take, every time the ARES-C is called up.

If F1 is pressed then the display will show:

```
A:                               R 7.44
EE
WAITNG A CALL

MUTE  DIR  STB  ESC
```

"A:" Shows the card selected for recording or playing
"EE" Means that the EE/AUTO/TAPE selector is set to EE. If F1 "MUTE" was pressed, the input signals will be muted. "R 7.44" In this example shows the remaining recording time on the selected card. F3 "STB" (STAND BY) gives the possibility to the user, to connect the ARES-C to an ISDN line waiting for a call and simultaneously to go to the "EDIT" mode. In this situation when during editing a call is coming in, the ARES-C will start beeping. The EDL can be saved first and afterwards it becomes possible to go back to the transmission menu to answer to the call.

If F3 "DIAL A NUMBER" was pressed then the display will show:

```
ENTER CALL NUMBER
█
* STAND BY

CLR  RCL  ESC  STO
```

Once this is displayed then the ISDN telephone number to be called can be entered using the numerical keypad. As soon as the user starts to enter the number to be called then the F1 position will display CALL. Once the entire number to be called has been entered, then F1 "CALL" can be pressed and the following will be displayed:

```
ENTER CALL NUMBER

001643546789 █
* STAND BY

CALL  CLR  RCL  ESC  STO
```

Once CALL is pressed the display will indicate "CALLING" and once the connection is made then it will indicate ON LINE and the communication will begin. Immediately the tax will be indicated below the words STAND BY in local currency (depending on the country). In Switzerland this is indicated as Fr 0.00 and will immediately indicate an initial charge that will increase once connection is made. This information is sent back to the ARES-C from the local exchange so the rate and taxation is the actual rate being charged by the Phone Company.

Once the display shows "ON LINE", the DTMF generator becomes available by simply pushing the numbers on the numeric keypad.

The * DTMF tone is generated using the "SHIFT key together with the "1" key.
The # DTMF tone is generated using the "SHIFT key together with the "2" key.

If the message *STAND BY appears (note the *), this means that a data communication has been selected (eg G722). If the message STAND BY appears (note WITHOUT *), this means that a speech communication has been selected (eg A-Law).

If the number called is occupied the display will indicate "BUSY" and the user can press call again when he is ready. There are actually some 30 or more different messages that may be sent back, each will be displayed and are self-explanatory.

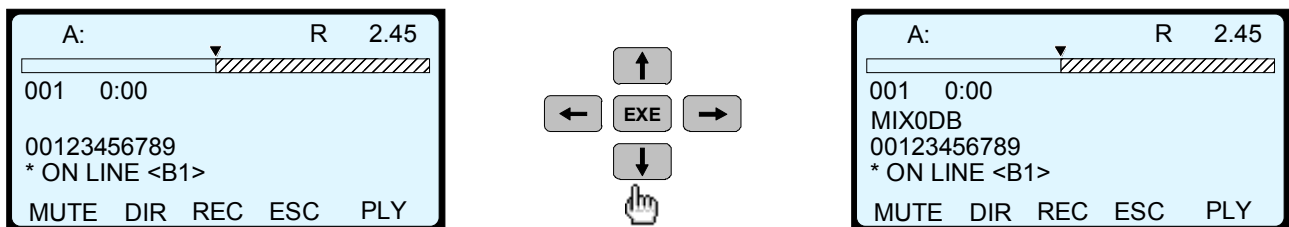
3.3.1 ISDN MIXING, DSP II

With the DSP II installed, it becomes possible to mix the playback signal with the line and or mic. inputs during an ISDN communication. This is possible for G.711, G.722 and MPEG ISDN communications at the moment that a playback selection inside the ISDN menu was made.

To enable this function, follow the steps below once the ISDN connection is established:

Important: It is not possible to switch on the mixing mode if the playback of the selected file was already started.

The next example shows an established ISDN connection before the play of the selected take was started.



A single push on the down arrow enables the mixing application with 0dB of playback attenuation.

Pushing several times on the down arrow will attenuate the playback signal in steps of 6dB (max. attenuation is -24dB)

Decreasing the attenuation is made by pushing the up arrow key. Increasing or decreasing the attenuation during the playback is possible.

Important: If the mixing option is not selected: the playback file stays first compression generation, if the mixing option is selected, even at 0dB, the playback file becomes second generation.

3.3.2 ISDN BIT-RATE CONVERSION, DSP II

If it happens during an ISDN transmission, that an MPEG stereo file or a different bit-rate file needs to be delivered as an MPEG file 64kb/s, follow the steps below:

Select the MPEG 64kb/s compression having the same sampling rate as the file that needs to be transmitted. Establish the MPEG ISDN connection and select the other bit-rate file. Playback the file. It will be automatically decompressed from Stereo to mono (128 or 192kb/s) and recompressed to MPEG 64kb/s before it goes on the ISDN line. It automatically becomes a second generation file.

3.3.3 ISDN COMPRESSION CONVERSION, DSP II

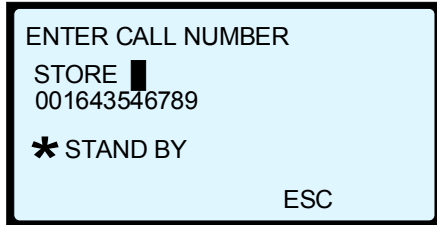
If it happens during an ISDN transmission, that a G.722 file needs to be delivered as an MPEG file, proceed as follows:

Select the MPEG 16kHz,64kb/s compression algorithm. Establish the MPEG ISDN connection and select the G.722 file. Playback the file. It will be automatically decompressed from G.722 and recompressed to MPEG before it goes on the ISDN line. It automatically becomes a second-generation file.

4.0 STORING / RECALLING INTERNAL TELEPHONE NUMBERS

The ARES-C has the possibility of memorising up to 10 regularly used telephone numbers. The numbers can be stored in the following manner:

Proceed as above as if a number were to be dialled manually, however instead of pressing F1 CALL to dial the number press F5 "STO" to store the presently displayed number in one of the memories 0 to 9 using the numeric keypad. When F5 is pressed the following display will be shown:



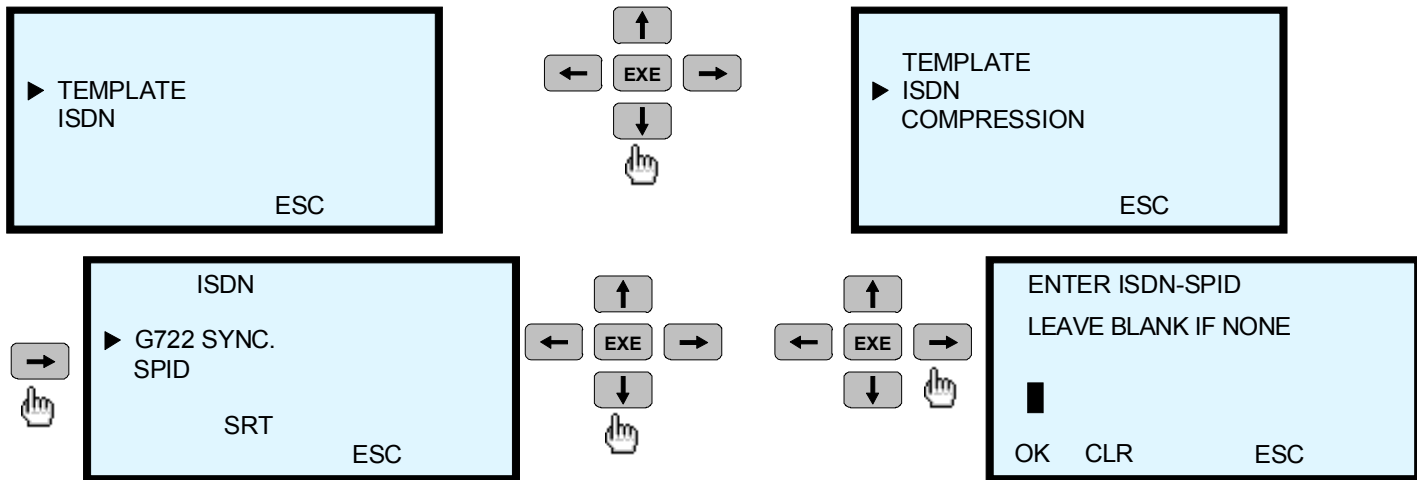
Simply press the memory number to be used.

In the reverse procedure, a number that has been previously stored in one of the memories can be retrieved by pressing RCL (recall) followed by the corresponding memory number (0-9). Finally the F2 key CLR is used to clear any information entered incorrectly.

5.0 SPID / USA APPLICATIONS

If an ISDN connection is made in the US using the NATIONAL-1 ISDN protocol, a Service Profile Identification number needs to be entered. The intention of this SPID is to allow the telephone exchange centre to automatically adapt the user requirements (voice, data, etc.).

From the main menu, select F3 "SET". The following submenu appears:



The SPID number can be set by using the numerical keyboard followed by F1 "OK" and is memorised in the RAM. As long as the battery box is not removed for a long period, the SPID stays in memory. The number can be erased by pushing F2 "CLR" if another SPID is needed. If no SPID is needed, proceed by pushing F1 "OK".

The rest of the procedure is identical as covered under para. 3.3.

The Telephone Company delivers the SPID number.

It includes the following information:

Example: 615 1234567 0100: 615 = area code, 1234567 = directory number, 0100 = identification

Pay attention that a SPID is dedicated to one "B" channel

A "BRI" ISDN connection contains 2 "B" channels which means 2 different SPID numbers.

6.0 COMPRESSION TABLE

Compression	Bitrate	Sampling frequency	Approx. bandwidth	AD converter	Compression level / channel
A-law	64kb/s	8kHz	3.5kHz	16 bit	2
μ-law	64kb/s	8kHz	3.5kHz	16 bit	2
G.722	64kb/s	16kHz	7.5kHz	16 bit	4
MPEG mono	64kb/s	16kHz	7.5kHz	16 bit	4
MPEG mono	64kb/s	24kHz	10.5kHz	16 bit	6
MPEG mono	64kb/s	32kHz	10.5kHz	16 bit	8
MPEG mono	64kb/s	48kHz	10.5kHz	16 bit	12
MPEG mono	128kb/s	16kHz	7.5kHz	16 bit	2
MPEG mono	128kb/s	24kHz	11.5kHz	16 bit	3
MPEG mono	128kb/s	32kHz	15kHz	16 bit	4
MPEG mono	128kb/s	48kHz	20kHz	16 bit	6
MPEG joint stereo	2x64kb/s	16kHz	7.5kHz	16 bit	4
MPEG joint stereo	2x64kb/s	24kHz	11.5kHz	16 bit	6
MPEG joint stereo	2x64kb/s	32kHz	15kHz	16 bit	8
MPEG joint stereo	2x64kb/s	48kHz	20kHz	16 bit	12
MPEG mono	192kb/s	32kHz	15kHz	16 bit	2.666
MPEG mono	192kb/s	48kHz	20kHz	16 bit	4
MPEG joint stereo	2x96kb/s	32kHz	15kHz	16 bit	5.333
MPEG joint stereo	2x96kb/s	48kHz	20kHz	16 bit	8

EXAMPLE: A-LAW is a sampling rate of 8kHz on 16 bits

$8000 \times 16 = 128000$ b/s, 128kb/s divided by 64kb/s = **2**

7.0 Which compression to be selected for best quality

The best solution is first of all to select the lowest compression ratio versus the requested bandwidth.

If no transmission is needed

For voice recording

Mono

If a bandwidth of 7.5 to 10.5 kHz is sufficient, MPEG mono 64kb/s 16 or 24kHz sampling should be used. This gives a compression ratio of "4 or 6" and when selecting a higher sampling rate, the bandwidth will not increase.

Stereo

If a bandwidth of 7.5 to 10.5 kHz is sufficient, MPEG joint stereo 2 x 64kb/s 16 or 24 kHz sampling should be used. This gives a compression rate of "4 or 6".

For music recording

Mono

If a bandwidth of 15 kHz or 20kHz is sufficient, MPEG mono 128kb/s 32 or 48 kHz sampling rate should be used. This gives a compression rate of "4 or 6". To reduce the compression rate, MPEG mono at 192kb/s 32 or 48kHz can be selected. This gives a compression ratio of "2.666 or 4"

Stereo

If a bandwidth of 15 or 20 kHz is sufficient, MPEG joint stereo 2 x 64kb/s 32 or 48 kHz sampling rate should be used. This gives a compression rate of "8 or 12". To reduce the compression rate, MPEG joint stereo at 2x96kb/s 32 or 48kHz can be selected. This gives a compression ratio of "5.333 or 8"

If transmission is needed

Transmission is only possible on a single "B" channel. This means that only 64kb/s can be transmitted.

Transmission to a standard telephone

The selection of compression in this case is limited to A-law or U-law (EUROPE - USA). Only a bandwidth of 3.5 kHz can be obtained.

Transmission to a codec

It's up to the codec side to decide for the highest common compression that can be selected. Not all codecs on the market are able to recover for a single 64 kb/s channel the possibilities of the ARES-C.

8.0 CODEC COMPATIBILITY

"1" means "AVAILABLE"

"0" means "NOT AVAILABLE"

Company	Model	64 kb/s	56 kb/s	EUR -I	TR 6	RT & 5	TT E	SS	NORTH AMERICAN DISDN-1	ALAW	μLAW	G.722	MPEG LA YER II	SAMP LING 16 KHZ	SAMP LING 24 KHZ	SAMP LING 32 KHZ	SAMP LING 48 KHZ	MPEG STEREO	2B ISDN CALLING	G.722 SYNC. SRT	G.722 SYNC. H221	G.722 SYNC. H242	SYNC. H221 + H242	MPEG SYNC J52	MONO 2X 64 K B/S	MSN		
TELOS	ZEPHYR	1	1	1	0	1	1	1	1	1	1	1	0	0	1	1	1	1	1	0	1	0	0	0	1	1	1	*3*
COMREX	DXR1	1	1			1				0	0	1	0	0	0	0	0	0	0	0	0	1			0	0	0	
DIALOG 4	MUSICTAXI L3	1	1	1	1	0	0	0	1	0	0	1	1	0	0	1	1	1	1	1	0	1	1	0				
VORTEX	VX-1000	1	1	1	0	0	0	0	0	0	0	1	1	1	1	1	1	0	0	0	0	1	0	0		0		
PHILIPS	7-KHZ-PKI	1		1	1	0	0	0	1	0	1	0	0	0	0	0	0	0	0	0	0	0	0	1	0	0	0	*1*
ACAMAS	REPAC	1		1		0	0	0	1		1	0	0	0	0	0	0	0	0	0	1		1	0	0	0		*2*
AETA	HIFISCOOP	1	1	1		0	0	0	0	0	0	1	1					1	1	0	1	1	1	1	1	1		
CCS	CDQ 1000	1	1	1						0	0	1	1	0	1	0	1	0	0	0	0	1	1	0		1	0	
CCS	CDQ 2000	1	1	1						1		0	1	0	1	0	1	1	1	1	0	0	0	0	1	1	1	
CCS	PRIMA 220	1	1	1							1	1	1	1	1	1	1	1	1	0	1	1			1	1	1	
TELECOM	YOU COM	1	1	1	1	0	0	0	1		1	1	0	1	0	1	0	1			1	1	0	1	1	0	1	
GLENSOUND	GSGC5	1	0	1						0	0	1	0	0	0	0	0	0	0	0	0	1	0	0	0	0	0	
NAGRA	ARES	1	0	1	0	1	1	1	1	1	1	1	1	1	1	1	1	0	0	1	1	1	1	1	0	0	0	0

1 The Philips 7kHz telephone in A-law mode works at a bit rate of 64kb/s, in G.722, it works in 56kb/s for the audio data

2 The REPAC in G.722, H221 mode works with 56kb/s audio data

3 The Telos Zephyr in MPEG II is not handling 32kHz sampling rate, except in MPEGIII

ATTENTION:

THE ARES-C with a DSP type I, IN MPEG COMMUNICATION CAN ONLY EMIT OR RECEIVE MPEG IN ONE DIRECTION, THE OTHER DIRECTION IS A-LAW.

THE ARES-C with a DSP type II, IN MPEG COMMUNICATION CAN EMIT OR RECEIVE MPEG IN BOTH DIRECTIONS and can also be set for MPEG-G.722 or MPEG-G.711 A-law.

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1.0 MAINTENANCE

1.1 Cleaning of the ARES-C

As the ARES-C has no moving parts there is virtually no maintenance required. There are no mechanical alignments to be verified. Our only recommendation is periodic cleaning of the exterior of the machine. For this purpose we suggest a soft cloth and Isopropyl alcohol. Never use chemicals such as Trichloroethylene as this may damage plastic parts of the recorder.

All electronic schematics and circuit lay-outs are available for the ARES-C and its accessories and can be found in the service manual.

2.0 PROBLEM SOLVER

This section is aimed to help the user to quickly identify and correct problems that may occur due to incorrect settings of the ARES-C. The problems are listed under different sections:

- Power
- Record
- Playback
- Monitoring
- Transmission
- Miscellaneous

If trouble is experienced with operating the ARES-C then a quick glance at the corresponding section should help.

2.1 POWER

2.1.1 Machine will not power ON

Check condition of batteries if dry cells are used.

Check charge of NiCds if used.

Check that the batteries are inserted with correct polarity. (+ve terminal to the right-hand side when viewed from the front of the machine)

Check that the main function switch was in the STOP position when the batteries were replaced.

2.1.2 Batteries go flat during record / editing

Automatic save will be performed before power OFF occurs. If editing then the work up to this point is saved with a new take number.

2.2 RECORD Machine will not record.....(No red leds alight)

2.2.1 No record at all

Make sure that the PCMCIA card is not set to Write Protect. Indicated by WP on the deck plate display next to the card.

Ensure that there is some space on the PCMCIA card(s). If cards are full either delete some takes or replace the card.

Check that the internal editor is OFF. (The ARES-C will not go into record mode if the editor is ON)

2.2.2 Microphone input

Check correct microphone powering is selected on the left side panel.
Check microphone sensitivity selectors on the front panel are correctly set.
Check that the microphone level potentiometers are not in their fully counter clockwise position if the ALC is turned OFF.
Check connection of the microphone cable (especially in "T" power operation).

2.2.3 Aux line input

Check that AUX is selected in the INPUTS menu mode.
Check that LINE POT menu is selected to LINE IN and that the Aux IN and Line out pot is not in the fully counter clockwise position.
Check that signal is fed to the correct pins of the 15-pin miniature "D" tape connector on the left side of the machine.

2.3 PLAYBACK

2.3.1 No playback at all (green leds OFF)

Check that the internal editor is OFF. (The ARES-C will not playback using the main function selector if the editor is turned ON)
Check that the PCMCIA card(s) have at least one take on them that can be played back.

2.4 MONITORING

2.4.1 No audio in headphones in playback

Check that the headphone level pot is not in the fully counter clockwise position.
Check that the Aux IN and Line out pot is not in the fully counter clockwise position.
Check that the Aux IN and Line out potentiometer is set to LINE OUT in the Line Pot Menu.
Check that the EE / AUTO / TAPE selector on the front panel is not in the EE position.

2.4.2 No loudspeaker output

Check that the headphone level pot is not in the fully counter clockwise position.
Check that the Aux IN and Line out pot is not in the fully counter clockwise position.
Check that the Aux IN and Line out potentiometer is set to LINE OUT in the Line Pot Menu.
Check that the EE / AUTO / TAPE selector on the front panel is not in the EE position.
Check that the speaker selection in the LOUD SPK menu is set to ON.

2.5 METERING No modulometer indication

2.5.1 Microphone connected machine In Test / Record mode...

Check microphone powering selection.

Check that the internal editor is OFF

Check that the modulometer is set to either AUTO or LEV IN in the modulometer menu.

Check microphone is connected to the correct input and that the microphone powering selectors are correctly set.

Check that the modulometer is not set to monitor the wrong channel (stereo mode only) with only 1 mic connected.

2.5.2 In playback...

Check modulometer is selected to LINE OUT or AUTO.

Check compression mode is the same as the recording on the card.

Check that the AUX IN and LINE OUT potentiometer is not set to the fully counter clockwise position.

Check that the TAPE / AUTO / EE selector is not in the EE position.

2.6 ERROR CODES

2.6.1 On front display

Error 1

Time out during a status of the ARES-C. This can happen for example if the machine goes from "STOP" to "PLAY". Mostly if Error 1 appears, the DSP is lost. Switching OFF and back ON the machine solves the problem.

Error 2

Data buffer is full. This can happen during "PLAY" or "RECORD". Switching OFF and back ON the machine solves the problem.

Error 3

DSP is not responding anymore at the start of a record. The red leds turn off immediately.

2.6.2 On top display during transmission.

No.	Display	Cause
1	"INVALID CALL NUMBER"	Unallocated (unassigned) number
2		No route to specified transit network.
3		No route to destination.
6		Channel unacceptable
7		Call awarded and being delivered in an established channel.
16		Normal clearing

No.	Display	Cause
17	= "BUSY"	User busy
18	= "NO ANSWER"	No user responding
19	= "NO ANSWER (ALERTED)"	No answer from user (user alerted).
21	= "CALL REFUSED"	Call rejected
22		Number changed
26		Non-selected user clearing
27		Destination out of order
28		Invalid number format
29		Facility rejected
30		Response to status enquiry
31		Normal, unspecified
34	= "NO FREE CHANNEL"	No circuit / channel available
38		Network out of order
41		Temporary failure
42		Switching equipment congestion
43		Access information discarded
44		Requested circuit / channel not available
47		Resources unavailable, unspecified
49		Quality of service unavailable
50		Requested facility not subscribed
57		Bearer capability not authorized
58		Bearer capability not presently available
63		Service or option not available, unspecified
65		Bearer capability not implemented
66		Channel type not implemented
69		Requested facility not implemented
70		Only restricted digital information bearer
79		Service or option not implemented, unspecified

No.	Display	Cause
81		Invalid call reference value
82		Identified channel does not exist
83		A suspended call exists, but call identity does not
84		Call identity in use
85		No call suspended
86		Call with requested call identity has been cleared
88	= "INCOMPATIBLE DEST."	Incompatible destination
91		Invalid transit network selection
95		Invalid message, unspecified
97		Message type non-existent or not implemented
98		Message not compatible with call state or message type non-existent or not implemented
99		Information element non-existent or not implemented
100		Invalid information element contents
101		Message not compatible with call state
102		Recovery on timer expiry
111		Protocol error, unspecified
127		Inter working, unspecified
139	= "NO NETWORK"	

2.7 MISCELLANEOUS

2.7.1 Card removed during formatting

Re-install card, main selector to EDIT /STD.BY. Press F4 "ON" followed by F4 "MIS", "FOR" and YES. This will reformat the card.

DECLARATION DE CONFORMITE DECLARATION OF CONFORMITY



FABRICANT: NAGRAVISION S.A. KUDELSKI GROUP, 1033 Cheseaux SUISSE
MANUFACTURER: NAGRAVISION S.A. KUDELSKI GROUP, 1033 CHESEAUX,
SWITZERLAND

APPAREIL : ARES-C
MODEL: ARES-C

NORMES GENERIQUES APPLICABLES :
APPLICABLE GENERIC NORMS:

EN 50081-1 (92) pour les émissions
For Emissions
EN 50082-1 (92) pour l'immunité
For Immunity

Par la présente nous déclarons l'équipement conforme aux exigences de protection de la directive européenne 89/336/CEE relative à la compatibilité électromagnétique pour environnement commercial et l'industrie légère.
We hereby declare that the equipment conforms to the requirements of the European guidelines 89/336/CEE referring to the electromagnetic compatibility for commerce and light industry.

Avertissement.

Cet appareil appartient à la classe A de la norme EN 50081-1 (92). Dans un environnement résidentiel, il peut provoquer des brouillages radioélectriques. Dans ce cas, il peut être demandé à l'utilisateur de prendre des mesures appropriées.

Warning.

This unit falls within the Class A of the norm EN 50081-1 (92). In a residential area it may cause radio interference. In this event the user may be required to take the necessary precautions.

Cheseaux 1^{er} trimestre 2002
Cheseaux 1st quarter 2002

APPENDICE 1

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1.0 MICROPHONES

The ARES-C is fitted with internal switchable microphone pre-amplifiers. The microphone inputs are on XLR female connectors on the left-hand side of the recorder. Located above the input connectors are two four position switches (with a slotted head) which allow the powering selection for the different possible microphone types. The possible selections are Dynamic, "T" +12 V power, Phantom +12 V and Phantom +48 V.

A microphone converts acoustic energy into an electrical signal. Numerous physical principles have been used to obtain this conversion and there are many different types of microphone available: Condenser, moving coil dynamic, ribbon dynamic, microphones etc. Neither the perfect nor the universal microphone exists. Each type has its defects and particular qualities, and the choice depends upon the required effect and particular application.

1.1 Sensitivity

Placed in a given acoustic field (e.g. Y hPa R.M.S.), a microphone will give a signal of X mV R.M.S. The ratio X/Y represents the sensitivity or, in effect, its efficiency. To give this value sense, it is also necessary to state the internal impedance of the microphone and the load impedance.

A classic dynamic microphone may have a sensitivity of 0.2 mV/hPa from 200 Ω internal impedance. A model giving 0.25 mV is considered to be sensitive, whereas a model giving 0.1 mV is unsuitable for capturing low level sounds.

Condenser microphones always have a preamplifier within the microphone casing, otherwise their high impedance would not allow the signal to be transmitted along a cable. At the output of the preamplifier a typical sensitivity figure is 1-4 mV/hPa with a load impedance of 200-1000 Ω . It is difficult to produce a very low noise preamplifier capable of receiving (without overloading) a signal given by a condenser microphone placed in a strong acoustic field (100 hPa). For this reason, it is better to have a special preamplifier for condenser microphones. The use of an attenuator between a condenser microphone and a preamplifier designed for a dynamic microphone is not recommended, as the signal-to-noise ratio will be unfavourable.

1.2 Frequency Response

The frequency response represents the sensitivity of the microphone as a function of the frequency. It is possible that the response will be different according to the direction from which the sound arrives into the microphone. This point is very important and will be dealt with in more detail later. Microphone manufacturers pay careful attention to the frequency response, and in general, most of the professional microphones available have a sufficiently good characteristic, at least for sounds arriving along the principal axis.

1.3 Coloration. Transient Reproduction. Reverberation

This shows that the frequency response, distortion and signal-to-noise ratio are not sufficient to describe an electro-acoustic device. A moving coil dynamic microphone makes use of resonances to render its frequency response flat. With continuous sinusoidal signals it functions perfectly, but when a signal appears suddenly, the resonating device needs a certain time to move. When a sound disappears suddenly, the resonator continues to produce a signal. The result is that the transient signal (e.g. a percussive sound) will be coloured by the inherent resonances of the microphones. This explains the difference noted by the ear between microphones with seemingly identical characteristics.

In general, condenser microphones use resonators only in the extreme high frequencies, where the coloration phenomena has little importance. As a result, their fidelity is excellent. Ribbon microphones can colour the low frequencies. Moving coil dynamic microphones colour to the greatest extent, this coloration is not always undesirable. They can improve certain voices, and the experienced engineer will not hesitate to use them under certain conditions. He can also use any defects in the frequency response for filtering, etc.

1.4 Use at High Sound Levels

Ribbon microphones and bidirectional condenser microphones can be damaged by a large air displacement. To record an explosion, a moving coil microphone, or better still, an omnidirectional condenser microphone is recommended. A switchable microphone (uni-, bi- or omnidirectional) risks the same damage as an ordinary bidirectional microphone. A microphone can be damaged under these conditions whether it is being used or not. It is advisable to place bidirectional and cardioid microphones in sealed boxes if an explosion is likely. Independent of the risk of damage, it is possible that a microphone will not reproduce well at levels greater than a certain value, above which the signal would become distorted. In general, moving coil microphones support the highest levels. Certain condenser microphones are designed so that an attenuator can be placed between the microphone capsule and the preamplifier.

1.5 Signal-to-noise Ratio

The recording of low level sounds can be disturbed by the combination of the microphone and its preamplifier. The word "combination" is used because the background noise does not come only from the amplifier. Take the case of a dynamic microphone whose impedance is $200\ \Omega$. As it does not have a temperature of absolute zero (-273°C) the electron movement in this impedance will produce a noise signal called the thermic noise. The preamplifier adds to the thermic noise its own inherent noise. The acoustic noise is measured in phons. The phons are decibels whose reference zero has been fixed by convention at $0.0002\ \text{hPa}$ (threshold of hearing). The measuring device is not linear, but has a frequency response similar to that of the ear. For low levels, this frequency response is called the ASA "A". It is possible to find out the equivalent acoustic noise level of a microphone and its preamplifier. Take for example a microphone of $200\ \Omega$ having a high sensitivity ($0.25\ \text{mV/hPa}$). Its noise level referred to the input will be $-126\ \text{dBm ASA "A"}$ (the dBm are decibels whose reference zero has been fixed at $1\ \text{mW}$). Now, $0.0002\ \text{hPa}$ is equivalent to $0.005\ \mu\text{V}$ ($139\ \text{dBm}$). Therefore the equivalent noise of this microphone will be $139 - 126 = 13$ phons.

This figure is correct only if the impedance of the microphone is $200\ \Omega$. Often, certain microphones whose nominal impedance is $200\ \Omega$ have higher impedances, at least in certain parts of the frequency spectrum. The effect of this is to increase the equivalent noise. A condenser microphone can also be characterized by an equivalent noise level, thereby making it possible to compare the performance of these microphones with that of dynamic ones.

1.6 Directional Characteristics

Often, when recording sound it is desirable to attenuate certain unwanted sounds, such as echoes coming from the studio walls. Certain microphones have a sensitivity which varies according to the direction from which the sounds come. In effect, these combine a pressure characteristic with a velocity characteristic. Taking into consideration the air pressure at any given point, a microphone acting as a manometer is called a pressure microphone. The direction from which the sound comes does not affect the pressure, except at very high frequencies, when the microphone makes its own shadow. On the other hand, the velocity of the air molecules can be used in a microphone. The word velocity implies a combination of speed and direction. A velocity microphone consists of a very light loose diaphragm which follows the displacement of the air. It will be sensitive to waves which strike the diaphragm perpendicularly whether they come from in front of, or behind it. Waves coming from the side will have no effect. This is the principle of velocity of bidirectional microphones. Such a microphone eliminates an important fraction of the reverberation and if the source of undesirable noise is well localized, it can be placed in the dead zone of the microphone. In combining a pressure microphone with a velocity microphone, an omnidirectional, or cardioid is obtained. The two elements are, of course, mounted in a common casing and electrically connected.

1.7 Secondary Characteristics Related to Directional Characteristics

Omnidirectional microphones (pressure) are much less affected by the wind than bidirectional (velocity) or cardioid microphones (because of their velocity element). The light diaphragms of velocity microphones have a tendency to float in the wind. It has been shown that the velocity microphones are easily damaged by a sudden air displacement (explosion). The response curve of an omnidirectional microphone is reasonably independent of the direction. However, sounds coming from behind will have a tendency to become muffled. Bidirectional microphones attenuate the lateral sounds in a relatively uniform manner, but cardioid microphones, and above all, dynamic ones, can have a very bad frequency response in the null directions. In other words, the attenuation varies greatly according to the frequency. If a cardioid microphone is used to eliminate undesired noises, this phenomena is not of great importance. If such a microphone is used to balance the sound, when a very loud source is placed around the null area of the microphone, it is advisable to check the results. The internal impedance of omnidirectional dynamic microphones is reasonably constant. They can therefore be used to feed their preamplifier either by voltage or current. On the other hand, the majority of cardioid microphones have an impedance varying greatly with the frequency. In this case only a voltage feed is recommended. Directional microphones only function well if they are sufficiently far from other objects which can disturb the acoustic field, because an obstacle disturbs the pressure less than the velocity.

1.8 Practical Advice on the Choice of the Microphones

1.8.1 Omnidirectional Microphones (pressure)

Robust, with low sensitivity to the wind, reproducing ambient sounds well-their price is lower than that of directional.

Principal Use: reporting

Special Uses: Lavalier microphone. For this use, special units have been created whose frequency response compensates for the perturbation of the body, and which takes into account the very low frequency sounds radiated directly from the chest. Recording music in the open air. Reverberation is non-existent and there are good microphones available - also very robust of low sensitivity, 0.1 mVhPa, which is acceptable as the sound level is reasonably high in these cases. Recording when the microphone is placed in the middle of a sound source (e.g. in the middle of an orchestra).

1.8.2 Bidirectional Microphones (velocity)

These give a very good attenuation of reverberation, and a good fidelity for sounds coming from the null direction. They are very sensitive to wind noise, and they accentuate the low frequencies if the sound source is very close. This phenomenon gives a very "Warm" effect, which is exploited by certain "charm" singers. Principal uses: music. Dialogue in the case where the microphone is placed between two speakers. Remarks: Dynamic bidirectional microphones, i.e. ribbon microphones, are either of very low sensitivity, or very bulky. Condenser microphones have a normal sensitivity.

1.8.3 Switchable Microphones

Certain condenser microphones can function as omni-, bi- or unidirectional by means of a simple switching.

1.8.4 Choice between Condenser or Dynamic Microphones

Condenser microphones give the best fidelity. In particular their reproduction of transient noises is excellent, but they cost more and are less robust than the dynamic microphones. They require a power supply either from the recorder or from an auxiliary device. They exist in two types: D.C. polarization and H.F. polarization. The performance and reliability depend, in the long run, more on the competence of the manufacturer than on the chosen system. Dynamic microphones are reputed to be more robust, but here again, the technological level of the manufacturer seems to be more important than the chosen system. The coloration which certain moving coil microphones give can be used to an advantage.

1.9 Maximum Gain of the Recording Chain or Sensitivity of the Microphone Inputs

In the case where a loud sound is recorded, the noise level is that of the quantizing noise of the A/D converter, the microphone noise level being lower, due to the reduced gain of the analog pre-amplifier chain. In these conditions, it may be useful to use a high recording level so that the signal-to-noise ratio is as great as possible. In the case where the sound source level is very low, the gain has to be increased to a point where the microphone/preamplifier combination noise level becomes more important than the quantization noise. Under these conditions, no advantage is obtained by recording near 0dB level. For these reasons, the sensitivity of the microphone preamplifiers has been limited under normal conditions to 0.2 mV into 200 Ω to enable a recording to be made at 0 dB.

1.10 Microphone sensitivity selection on the ARES-C

The sensitivity of the microphone pre-amplifiers on the ARES-C are selected by means of the two switches #6 marked "Sensitivity" on the front panel of the machine located below the input potentiometers. Three possible selections can be made for each microphone input these being 1mV/hPa, 4mV/hPa and 0.2 mV/hPa. (see also under input potentiometer)

1.11 Microphone powering

The selection of the different powering possibilities of the ARES-C is made using the two four position selectors located above the input connectors (with a slotted head) which allow the powering selection for the different possible microphone types. The possible selections are Dynamic, Condenser +12 V "T" power, Phantom +12 V and Phantom +48 V. With this new concept of powering independent from the sensitivity selection, the user is free to make the required settings depending upon the nature of the recording being made.

2.0 METERING

To measure the level of an electrical signal representing a sound, there are two devices available, the modulometer and the VU meter. Both of them are voltmeters whose needle position represents the level. Their construction and uses are however different.

2.1 Modulometer

The modulometer measures the peak value of the signal, irrespective of the form or the level, the modulometer takes into consideration the strongest positive or negative value. It is equipped with a memory, so the signal can be very brief, but the memory ensures that the meter needle advances and stays there for sufficient time for the operator to read it. The essential advantage of the modulometer comes from the fact that the measurement it gives is that which directly concerns any system that can saturate, in the case of the ARES-C it is the signal peak which saturates the A/D converter. The average value of the signal (as far as the listener is concerned) is of no importance to the A/D converter. In particular, while recording short pulses, the modulometer indication is always exact, no matter how long the duration of the pulses. The norm stipulates that a pulse of 10 mS at 0 dB should indicate -2dB on the modulometer.

2.2 VU Meter

In the days of electronic valves (tubes), a modulometer was very costly, and the rudimentary VU meter was often preferred. Later, it was noticed that the VU meter still maintained a certain following and because of habit and standards many radio stations still use them.

A VU meter is a simple rectifier voltmeter whose response time has been standardized and indicates the physiological effect. If the signal to be measured is continuous, (e.g. a whistle) the VU meter will indicate a value the same as the modulometer, but if the signal is intermittent (e.g. speech) the VU meter will only indicate an average value, i.e. considerably lower than the instantaneous maximum levels. For speech, it has been found that this average value is approximately 8 dB lower than the peak value. By increasing the VU meter sensitivity by 8 dB, an indication of 0 VU is obtained when the peaks reach the maximum value. This works relatively well in practice. For noises, the indication of the VU meter evidently becomes very inexact, and renders it practically useless.

The VU meter, however, has certain advantages:

- a) **Speech-music balance.** If speech and music are recorded with a modulometer so that the peaks of the signal do not exceed the maximum level, subjectively the music appears stronger. This is due to the more continuous character of music signals. Therefore, in a mixed programme, it is necessary to modulate the speech more strongly than the music. This can be done by modulating the music correctly and overmodulating the speech or by undermodulating the music.
It is to be noted that a slight overmodulation of speech is not catastrophic: a transmitter is fitted with a limiter, which cuts peaks exceeding the maximum level. The subjective deterioration of the sound quality remains unnoticeable.
A VU meter under indicates with speech. In modulating a programme to 0 VU the speech will be overmodulated and the music under modulated. From this point of view, the VU meter seems to be of more interest for mixed transmissions whose quality is not of great importance.
- b) **The VU meter has a non logarithmic scale.** For the needle to move, it is necessary for the signal level to exceed -20 dB. This causes the operator to compress more than is necessary, i.e. to increase the level of the pianissimo. This reduces the quality of a musical transmission while increasing the range. On the other hand, it is favourable if the listener is in a noisy ambience, such a car or a cinema hall.

The meter on the ARES-C is a microprocessor controlled peak meter with a reaction time programmed in such a manner as to simulate the movement of a modulometer. Being microprocessor controlled gives several advantages over conventional meters. First, any desired ballistics can be used making it very flexible and secondly maximum levels can be stored in memory and recalled at the users discretion. The scale of the meter on the ARES-C ranges from -30dB to + 9dB. It should however be noted that although an indication of greater than 0 dB can be displayed on the input, the A/D will be saturated above 0 dB. The meter of the ARES-C can be selected to monitor either the input signal or the output signal and it is important to be sure that it is indicating the input during recording so that saturation of the A/D converter can be avoided.

3.0 LEVELS

Control of the Input Sensitivity (Modulation) Dynamic range, Signal-to-noise ratio, Decibels

The dynamic range is the ratio between the loudest and softest sound levels. The dynamic range is large for a symphony orchestra compared to that of an announcer reading a news bulletin.

The signal-to-noise ratio is related to the dynamic range. It is important that the softest sound level to be recorded is considerably stronger than the noise. Thus sound with a large dynamic range requires a high signal-to-noise ratio. However, this ratio can be practically equal to the dynamic range in the case where the noise level is close to the threshold of audibility. The subjective perception of the sound level follows a law which is approximately logarithmic. It is for this reason that it is customary to measure sound level as a logarithmic unit. This is the decibel (dB). Each time the sound power is multiplied by 10, the number of decibels which that represents is increased by 10. Thus an increase of 100 times equals 20 dB, a 1000 times equals 30 dB etc. It should be remembered that the power is proportional to the square of the amplitude. The voltage which a microphone gives is proportional to the amplitude. In other words, if the voltage increases 10 times, the power increases 100 times and corresponds to 20 dB. The decibel is a measure of power ratio and not an absolute value. In taking as a reference, a sound corresponding to a variation of pressure of 2×10^{-4} μ bar (value considered as the threshold of audibility at 1 kHz) a scale in absolute value will be obtained. A sound of 90 dB will therefore mean 90 dB above 2×10^{-4} μ bar. The sensitivity of the human ear varies with the frequency. In order to compensate for this, the sound level should be measured with filters simulating the variations of sensitivity of the ear. Thus the decibels become the phon referred to 2×10^{-4} μ bar.

The potentiometer scales of the ARES-C are graduated in decibels referred to 2×10^{-5} Pa for 0 dB.

$1 \text{ Pa} = 1 \text{ N/m}^2 = 10^{-5} \text{ bar}$ (Pa = Pascal, N = Newton)

$1 \text{ h} = 10^2$ (h = hecto)

$2 \times 10^{-4} \mu\text{bar} = 2 \times 10^{-5} \text{ hPa}$

At 1 kHz, these decibels are the same as phons but as the ARES-C does not have such filters, it cannot be considered as a phon meter. With the sensitivity potentiometer adjusted for 70 dB, a sound of 70 dB captured by a normal microphone (0.2 mV/ μ bar into 200 Ω) and attacking a normal sensitivity preamplifier, produces a recording at -10 dB level. The modulometer will indicate -10 dB.

At 125 dB, the scale is divided in two parts: The area from infinity to 125 dB is considered as a danger for overloading the preamplifiers and is only a zone for fading-out. This means that the audio signal is too strong. It is therefore necessary either to reduce the signal level by means of an attenuator, provided the microphone itself is not saturated, or to switch the input to a less sensitive position.

The area from 125 dB to 74 dB is the range that needs to be used for a correct 0 dB recording keeping in mind that close to the 74dB area, the preamplifier noise, together with the noise depending on the type of compression selected, becomes more important.

3.1 Input sensitivity on the ARES-C

The microphone sensitivity of the ARES-C can be switched as explained earlier to one of three positions according to the microphone being used these are 1mV/hPa, 4mV/hPa and 0.2 mV/hPa

If we take the last case, of 0.2 mV/hPa and we then turn the input potentiometer to maximum and we arrive with 0 dB on the meter of the machine this corresponds to an acoustic level of 74 dB. It is for this reason that the scale of the input potentiometer stops at 74 dB.

Other important indications are given on the scale of the input potentiometers that should be explained. The black mark on the scale of the ARES-C that starts at the 125 dB point is an important indication. If the potentiometer needs to be below this point (between 125 and ∞) for the meter to indicate 0 dB then this indicates that the input signal is so strong that the microphone pre-amplifiers are overloaded. Above the 125 dB mark indicating 0 dB on the meter will not cause the input stages to overload.

4.0 ALC (Automatic Level Control)

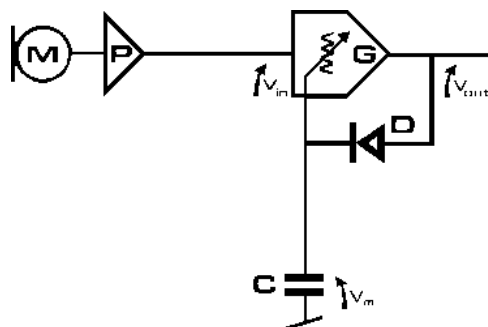
A field reporter using a tape recorder is obliged to take care of the interviewee, handle the microphone and at the same time, adjust input level to ensure good reproduction of his recording.

Thus, in order to alleviate the operational tasks and thereby enable the reporter to concentrate more on the interview itself, the NAGRA ARES-C is fitted with an ALC device, which can be turned on by the user. This internal accessory automatically adapts the sound level to a nominal recording level. In other words, an insufficient sound level is amplified, whereas loud sounds are attenuated.

The following explanation is based on the processing of an analog signal, part of this processing is undertaken by the microprocessor in the ARES-C.

Operation theory of an ALC

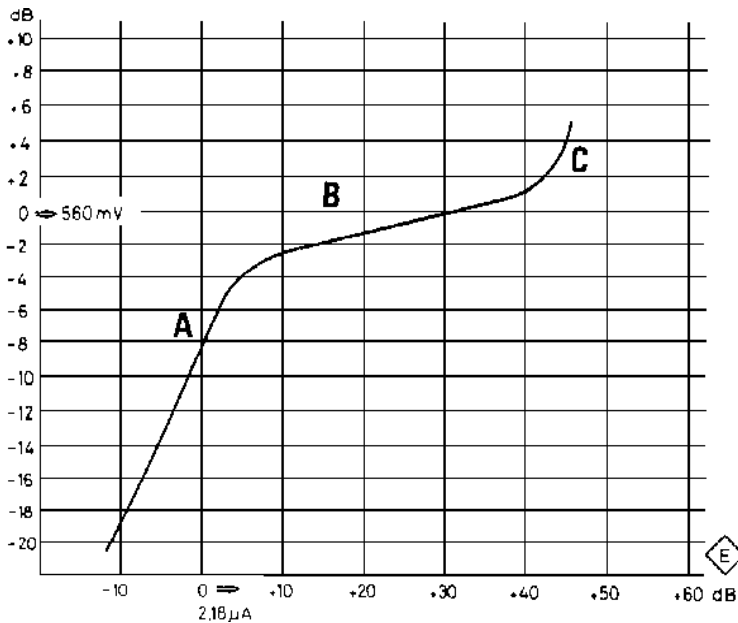
The signal from microphone M is first amplified by a low-noise pre-amplifier P which has a high dynamic range. The latter must be able to amplify without distortion all signals from the microphone (for an average sensitivity dynamic microphone, i.e. 0.2 mV/hPa, signal amplitude can reach 100 mV).



In the following amplifier G, the gain is adjusted by DC voltage. Then the corrected amplified signal charges the memory condenser C with a V_m voltage proportional to the output voltage V_{out} .

The voltage V_m is used to vary the gain of amplifier G which will decrease as V_m increases.

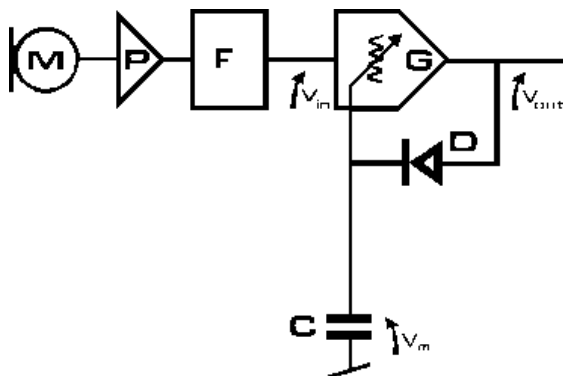
This constitutes a very simple ALC. The compression characteristics of such a device are shown below:



The linear part B corresponds to the zone where the ALC is operating. It will be observed that an input signal variation of 35 dB is converted to an output signal variation of only 3 dB. The linear part A shows that for an input signal variation of 20 dB we also obtain a 20 dB variation at the output, from which can be deduced that the amplifier gain is constant in this zone. The part C corresponds to the appearance of saturation of the adjustment unit (diode, condenser, amplifier).

High pass filter requirement

The ALC system described above is still very incomplete. To effectively carry out its task, several accessories are required, firstly a high pass filter F.



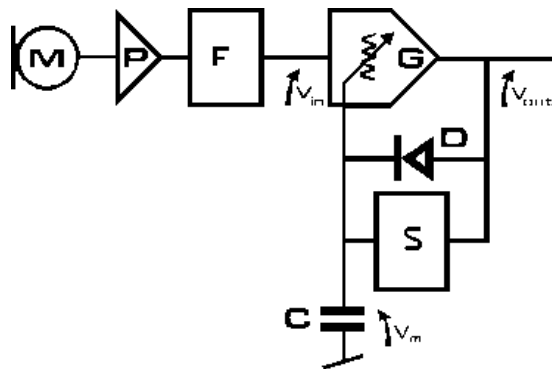
This filter is positioned before the variable gain amplifier G. Indeed, if a recording is made where the level of infrasound is higher than the useful signal (e.g.) in a car or near an open window), the adjustment of amplifier gain would be influenced and the useful signal would "die away".

To summarize, it can be said that it is most important that upon arrival at the amplifier G, the input signal no longer includes very low frequency signals. At high frequencies, this phenomenon does not occur due to the low level of signals above 8 kHz. Filter F characteristics are shown below: (if "FLAT" is selected)

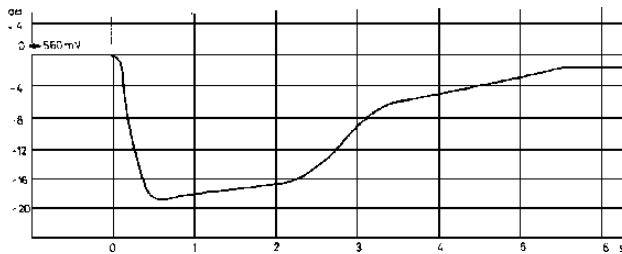
In our diagram, the condenser C constitutes a very rudimentary memory. For instance, if a recording is being made where signal amplitude varies a great deal, the condenser will memorize the largest correction required but neglect low amplitude signals. Such a situation arises when, for example, an interview begins in

a noisy place and without interrupting the discussion both parties move to quieter surroundings.

Thus, memory time must be limited by discharging the condenser and this is carried out by silence detector S which is activated when the absence of V_{out} signal lasts more than 2 seconds.

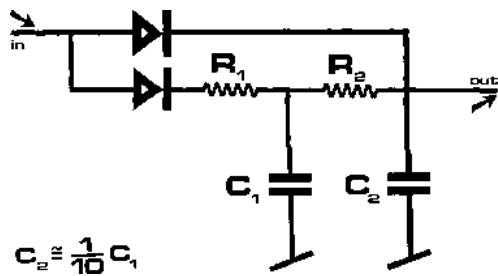


Following a sudden -20 dB decrease of the useful signal, the recuperation curve would be as follows:



The simple condenser memory described will, however, function quite differently depending on whether load constant is great or small. In the latter case, the memory takes into consideration the shortest pulses and memorizes them, meaning that throughout memory limitation phase following such a sound, recording level of the useful signal will remain "reduced". The A/D converter, however, will not receive any signal capable of saturating.

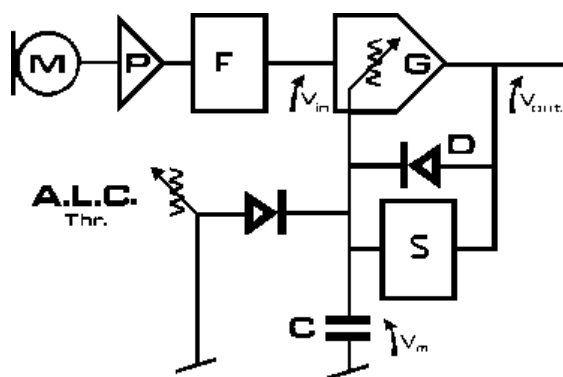
On the other hand, a greater load constant will integrate short signals and thereby saturate the A/D or recording circuits. By combining these two points, we have the real memory system of the ALC.



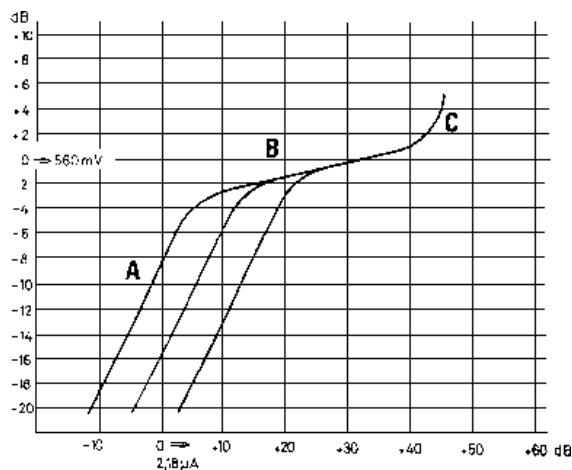
A very short signal at IN input quickly charges C2 and the output voltage immediately acts upon the variable gain amplifier. The same signal, integrated by R1 C1 has no effect on C1, but C2 will rapidly discharge through R2 into C1. Since $C_1 = 10 \times C_2$, the voltage at its terminals will be hardly affected. The whole operation takes a matter of milliseconds, but both the recording amplifier and the A/D have been protected against saturation.

At a more uniform recording level, C1 and C2 work in parallel. If an interview takes place where noise level is comparable to a restaurant or a busy street, etc., this will present a problem for the operator during long silences.

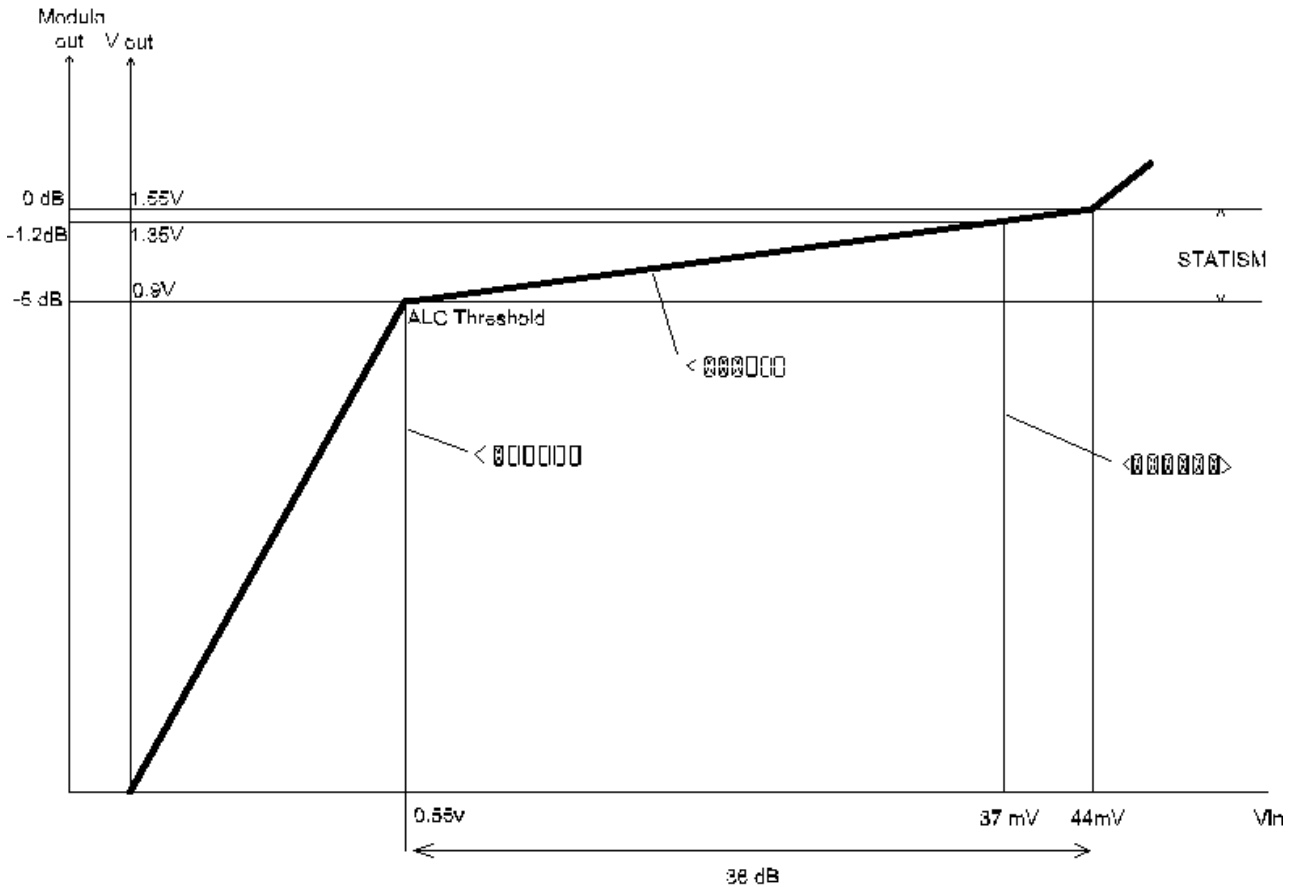
In the preceding paragraph, it has been shown that after 2 sec. silence, the silence detector will discharge the condenser until the useful signal is picked up again. But for the recording, this will be heard as an annoying increase of ambient noise level. So to avoid this effect, the operator has at his disposal control of the ALC. He can, then, select the point where compression begins and thus limit the active zone of the ALC. In this way, the ALC is used as limiter. This adjustment is selectable by menu.



This adjustment produces a compression curve as shown below:



In the ARES-C the statism of the ALC is selectable by menu as shown in the graphic below:



ALC of the ARES-C measured in the 4 mV/hPa position

The display of the ARES-C indicates as shown on the above graphic depending on the amount of compression when the ALC is active.

Each display icon on the front display of the machine corresponds to a threshold of 6 dB. The operating threshold of the ALC of the ARES-C can be set to one of four different levels. In the -18 dB position 3 icon squares will be alight permanently, in the -24 dB position 2 squares will be alight, in the -30 dB position 1 square will be alight and in the -36 dB position no squares will be alight.

The other setting that is possible for the ALC of the ARES-C is the speed of the ALC circuit. Three possible selections are possible, each corresponds to the "reaction" of the ALC. NORMAL is the standard operating position, which is similar to the operation of the ALC circuitry in the NAGRA IS and this corresponds to a signal hold of 2 seconds followed by a fall off of 6 seconds. The FAST position has a hold time of 0.4 seconds and a fall off during 1.2 seconds and the SLOW position has a hold of 2 seconds and a fall off of approximately 35 seconds.

APPENDICE 2

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1.0 ISDN BASICS.

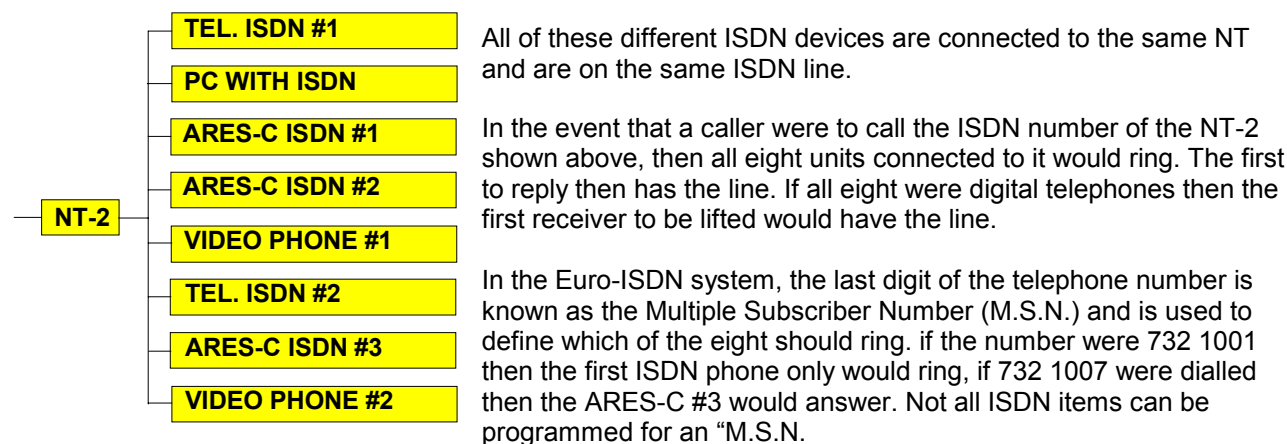
A standard analog telephone line consists of two copper wires running from the users telephone unit to the external telephone exchange and is more commonly known as a Public Switched Telephone Network or P.S.T.N. When the receiver of the analog telephone is lifted a relay closes and makes a circuit causing a current to flow through the telephone line, This current alerts the exchange to attribute a free line and waits for a series of numbers (either pulses or tones) to follow. This two wire system is more commonly known as a "U" connection. The acronym ISDN stands for Integrated Services Digital Network which is a digital telephone network using the same two copper wires but has to be connected to a digital exchange. The standard analog line is a two wire system and ISDN is a four wire system which is more commonly known as an "S" connection. In order to transform the users "U" connection (2 wire) into an "S" connection (4-wire) an interface known as an NT needs to be installed. This unit not only converts from 2 wire to 4 wire but also serves for line equalization.

The four wire system of ISDN consists of two symmetrical pairs known as TX-, RX-, TX+ and RX+. A single ISDN line is used to pass digital data and will permit 2 x 64 kbits and 1 x 16 kbits to be transmitted along it. These data streams are known as channels, so one ISDN line consists of 2 x "B" (bearer) channels and 1 x "D" (delta) channel.

If an ISDN line has been installed with the NT it can still be used in the same way as a standard telephone line providing that the telephone unit is a special digital telephone. If this is the case then the user can dial normally to a standard analog number. All information passing on an ISDN line is compressed therefore the A/D and D/A conversion as well as the compression and decompression is done within the special digital telephone unit. Such a digital telephone can be represented as follows:

If the handset is lifted on the digital phone, then immediately one of the "B" channels is used and part of the "D" channel. The "B" channel will carry the audio data and the "D" channel is used to carry communication control data such as the number being called, the return signal coming from the receiver with its number as well as the tariffs for the connection etc. As this system uses only one "B" channel, it is possible to connect a second digital telephone to the same ISDN line if required, and this will use the second "B" channel.

Actually there are two different types of "NT". The standard version is the NT-1 which allows the connection of two separate pieces of ISDN equipment to the line. The other type is the NT-2 which allows the connection of eight ISDN devices to the line. An NT-2 interface can be represented as follows:



2.0 ISDN SYSTEMS

The format of the Euro ISDN standard is as described above. That is to say 2 x "B" channels of 64 kbits/sec and 1 x "D" channel of 16 kbits/sec. A second system also exists, and is known as the "Switched 56" system. This consists of transmission of the signal at 56 kbits/second. This latter system is more commonly found in the U.S.A.. Most countries that are presently installing ISDN are opting for the Euro ISDN system.

The old European network called "1TR6" will not be implemented.

3.0 COMPRESSION USED IN THE ISDN DOMAIN.

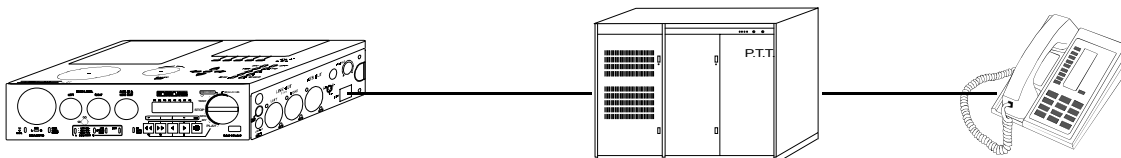
As was briefly mentioned earlier, the signals sent down the ISDN line are compressed. The compression system used may vary from country to country. In Europe, an ISDN telephone will normally use the A-Law compression standard which allows a bandwidth of 3.5 kHz. In the United States a similar telephone will generally use the μ -Law compression standard. The G722 compression standard however is also used on both sides of the Atlantic, and slowly the MPEG-1 Layer II is being installed.

Today, for example if the ARES-C is used to telephone from an ISDN line to an analog P.S.T.N. telephone from Cheseaux, then the ARES-C must be in the A-Law compression mode as our local exchange can presently only deal with A-Law signals. In this case the signal comes out of the ARES-C in the ISDN format compressed according to the A-Law algorithm and arrives in the local exchange. The signal is then de-compressed and fed through a D/A converter and then directed to the analog telephone line:

If however we were to use two ARES-C recorders as opposed to an ARES-C and an analog telephone then the DSP and the A/D in the central exchange are not needed as both connections are in "DATA" transmission mode. Hence the central exchange becomes transparent and the ISDN signal passes directly through it. In this case any of the compression modes of the ARES-C can be used as long as both machines are set to the same one before the transmission is made.

4.0 COMMUNICATION MODES. G.711-A (A-law), G.711-U (μ -law), MPEG (64kb/s)

4.1 A-LAW FULL DUPLEX.



Calling with the ARES-C from Europe to a standard telephone situated anywhere in the world.

Setup: On the deck, select in "SET", "COMPRESSION" the **G.711-A** mode.

Next steps: Put the ARES-C in EDIT/STD. BY position

- Switch on the editor on top
- Select F2 "TRM"
- Select F3 "ISDN"
- Select F2 if no take need to be pre-selected
- Select F3 if you need to call
- Enter the complete ISDN number
- Push F1 "CALL"

4.2 μ -LAW FULL DUPLEX.

Calling with the ARES-C from the USA to a standard telephone situated anywhere in the world.

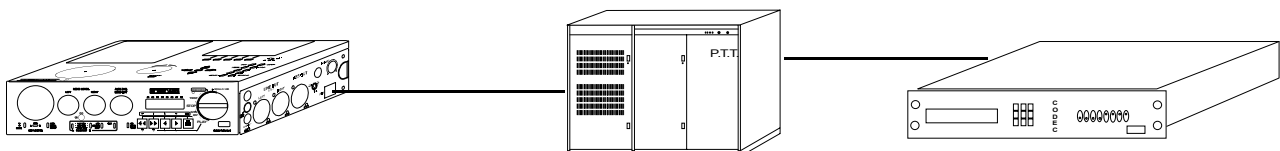
Setup: On the deck, select in "SET", "COMPRESSION" the **G.711-U** mode.

On the deck, select in "SET", "ISDN" the SPID submenu and enter the SPID number.

The next steps are the same as in paragraph 4.1.

4.3 A-LAW - μ -LAW FULL DUPLEX.

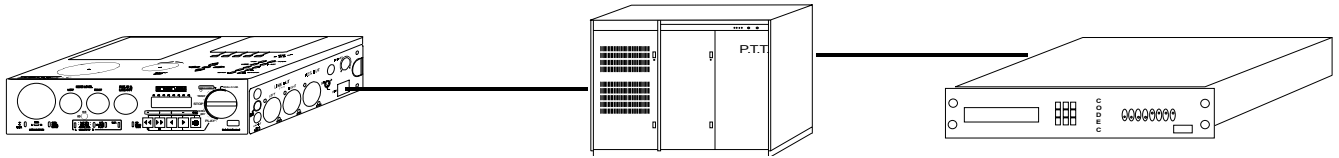
Calling with the ARES-C from Europe to an in the USA ISDN telephone or Codec.



Setup: On the deck, select in "SET", "COMPRESSION" the **G.711-A** mode.

The next steps are the same as in paragraph 4.1.

4.4 G.722 FULL DUPLEX.



Calling with the ARES-C from Europe to a Codec anywhere in the world.

Setup: On the deck, select in "SET", "COMPRESSION" the **G.722** mode.

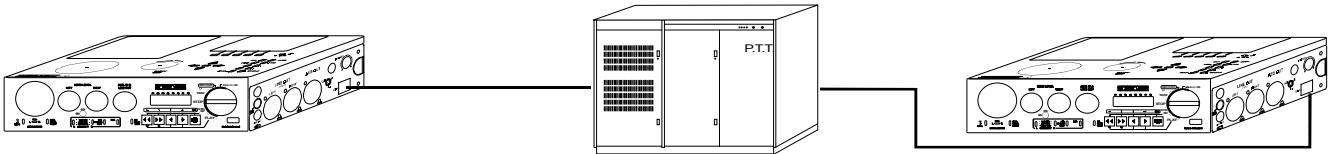
Sync. Protocol selection:

In the case of G.722, different sync. Protocols can be requested. Those are located on the deck in "SET", "ISDN", "G.722 SYNC." submenu.

For some French codecs, the "H221" protocol is requested. For some German codecs, the "H242" protocol is requested. In this case before making the ISDN contact, it is advised to verify which protocol is used. The most used protocol for the other codecs is "SRT".

Select the corresponding protocol.

The next steps are the same as in paragraph 4.1.



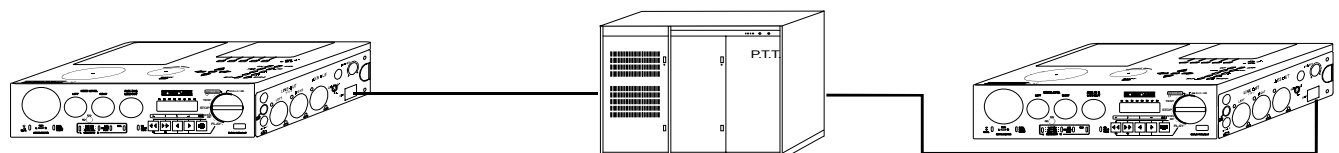
Calling with the ARES-C from Europe to an ARES-C anywhere in Europe.

Setup: On the front keyboard, select **G.722** in the menu.

Sync. Protocol selection:

In this case, any sync. can be selected as long as the other ARES-C has selected the same sync. Even the protocol "NONE" can be used.

4.5 MPEG - A-LAW FULL DUPLEX.



REPORTER SET

STUDIO SET

Calling with the ARES-C from Europe to an ARES-C anywhere in Europe.

Setup: On the deck in "SET", "COMPRESSION", select **MPEG, MONO, 64kb/s** followed by one of the four different sampling rates.

On the deck in "SET", "ISDN", "MPEG MODE", select **MPEG G711-A**.

Pay attention that both ARES-C's must have the same MPEG setup.

REPORTER SET Next steps: Put the ARES-C in EDIT/STD. BY position

Switch on the editor on top

Select F2 "TRM"

Select F3 "ISDN"

Select F2 if no take need to be preselected

Select F3 if you need to call

Enter the complete ISDN number

Push F1 "CALL"

In this case, the reporter set will transmit in the MPEG mode and receive in the A-LAW mode.

STUDIO SET Next steps: Put the ARES-C in EDIT/STD. BY position

Switch on the editor on top

Select F2 "TRM"

Select F3 "ISDN"

Select F3 "UPLOADING"

Select F1 "INCOMING CALL"

Push F1 "ANS" when the ARES-C starts ringing.

In this case, the studio set will transmit in the A-LAW mode and receive in the MPEG mode.

4.6 A-LAW - MPEG FULL DUPLEX.

Setup: This is the same as in the previous paragraph 4.6, except that the studio set and the reporter set are swapped.

4.7 MPEG - G.722 FULL DUPLEX.

Setup: This is almost the same setup as in the previous paragraph 4.6 but in the "SET", "ISDN", "MPEG MODE" submenu, **MPEG G722** is selected instead of MPEG G711-A.

4.8 G.722 - MPEG FULL DUPLEX.

Setup: This is the same as in the previous paragraph 4.7, except that the studio set and the reporter set are swapped.

4.9 MPEG FULL DUPLEX.

Setup: This is almost the same as in the previous paragraph 4.4, except that on the deck in "SET", "COMPRESSION", MPEG MONO 64kb/s is selected and in "SET", "ISDN", "MPEG MODE", **MPEG MPEG** is selected.

APPENDICE 3

1.0	FLASH CARDS COMPATIBILITY LINEAR AND STRATA	Page 2
2.0	FLASH CARDS COMPATIBILITY ATA AND COMPACT ATA	3

1.0 FLASH CARDS COMPATIBILITY, LINEAR AND STRATA.

TYPE		CAPACITY	MANUFACTOR	TECHNO.	IDENTIFICA.	ARES-C	ARES-P RCX220	ARES 95	ARES NT	ARES IMPORT	RCX LOAD
PCMCIA	1	20	INTEL	Linear	IMC020FLSA-15	Yes	Yes	Yes	Yes	Yes	Yes
PCMCIA	2	20	EDI	Linear	FLA2800C15	Yes	Yes	Yes	Yes	Yes	Yes
PCMCIA	3	10	INTEL	Linear	IMC010FLSA-15	Yes	Yes	Yes	Yes	Yes	Yes
PCMCIA	4	20	EDI	Linear	FLA3200C15	Yes	Yes	Yes	Yes	Yes	Yes
PCMCIA	5	20	EDI	Linear	FLA2400C15	Yes	Yes	Yes	Yes	Yes	Yes
PCMCIA	6	40	EDI	Linear	FLA3200C15	Yes	Yes	Yes	Yes	Yes	Yes
PCMCIA	7	64	EDI	Strata	FLF1200C25	Yes	Yes	Yes	Yes	Yes	Yes
PCMCIA	8	64	EDI	Strata	FLF1203C25	Yes	Yes	Yes	Yes	Yes	Yes
PCMCIA	9	128	EDI	Strata	FLF1203C25	Yes	Yes	No	No	No	Yes
PCMCIA	10	80	EDI	Strata	FLF0203C25	Yes	Yes	No	No	No	Yes
PCMCIA	11	192	EDI	Strata	FLF1203C25	Yes	Yes	No	No	No	Yes
PCMCIA	12	48	EDI	Strata	FLF0203C25	Yes	Yes	Yes	Yes	Yes	Yes
PCMCIA	13	64	EDI	Strata	FLF0203C25	Yes	Yes	Yes	Yes	Yes	Yes
PCMCIA	14	128	PRETEC	ATA	AFH128	Yes	Yes	No	No	Yes	Yes
PCMCIA	15	64	PRETEC	ATA	AFH064	Yes	Yes	No	No	Yes	Yes
PCMCIA	16	64	SANDISK	ATA	AB0120JR-USA	Yes	Yes	No	No	Yes	Yes
PCMCIA	17	48	SANDISK	ATA	V0004GW-USA	Yes	Yes	No	No	Yes	Yes
PCMCIA	18	64	EDI	ATA	ATA2500C25	Yes	Yes	No	No	Yes	Yes
PCMCIA	19	96	EDI	ATA	ATA2500C25	Yes	Yes	No	No	Yes	Yes
PCMCIA	20	128	EDI	ATA	ATA2500C25	Yes	Yes	No	No	Yes	Yes
PCMCIA	21	256	PRETEC	ATA	AFH256	Yes	Yes	No	No	Yes	Yes
PCMCIA	22	128	FEIYA	ATA	TS128MFLASHB	No	No	No	No	No	No
PCMCIA	23	48	PRETEC	ATA	AFH048	Yes	Yes	No	No	Yes	Yes
PCMCIA	24	96	PRETEC	ATA	AFH096	Yes	Yes	No	No	Yes	Yes
PCMCIA	25	80	CENTENIAL	Strata	ES00080	No	No	No	No	No	No

2.0 FLASH CARDS COMPATIBILITY, ATA AND COMPACT ATA.

TYPE		CAPACITY	MANUFACTOR	TECHNO.	IDENTIFICA.	ARES-C	ARES-P	ARES	ARES	ARES	RCX
							RCX220	95	NT	IMPORT	LOAD
COMPACT	C1	64	EMTEC	ATA	347629AI	No	No	No	No	No	No
COMPACT	C2	64	RIDATA	ATA	RITEK 06415006D	No	Yes	No	No	Yes	Yes
COMPACT	C3	128	ACE	ATA	FFC128	Yes	Yes	No	No	Yes	Yes
COMPACT	C4	64	SANDISK	ATA	AB0105LEI	Yes	Yes	No	No	Yes	Yes
COMPACT	C5	128	NAGRA(ACE)	ATA	FFC128	Yes	Yes	No	No	Yes	Yes
COMPACT	C6	64	ACE	ATA	FFC064	Yes	Yes	No	No	Yes	Yes
COMPACT	C7	64	NAGRA(ACE)	ATA	FFC064	Yes	Yes	No	No	Yes	Yes
COMPACT	C8	48	PRETEC	ATA	ACT048	Yes	Yes	No	No	Yes	Yes
COMPACT	C9	64	PRETEC	ATA	ACT064	Yes	Yes	No	No	Yes	Yes
COMPACT	C10	128	PRETEC	ATA	ACH128	Yes	Yes	No	No	Yes	Yes
COMPACT	C11	32	ACE	ATA	FFC032	Yes	Yes	No	No	Yes	Yes
COMPACT	C12	32	DATAFAB	ATA	922022134	Yes	Yes	No	No	Yes	Yes
COMPACT	C13	128	MEMORY CARD	ATA	TS128MFLASHCP	Yes	Yes	No	No	Yes	Yes
COMPACT	C14	48	SANDISK	ATA	V0006GP-CHINA	Yes	Yes	No	No	Yes	Yes
COMPACT	C15	256	PRETEC	ATA	ACH256	Yes	Yes	No	No	Yes	Yes
COMPACT	C16	96	PRETEC	ATA	ACH096	Yes	Yes	No	No	Yes	Yes

All ATA type & compact ATA type cards with this background color are accepted and sold by Nagra

AresImport test was done using an external PCMCIA SCSI adapter on NT-4

All cards went into the record mode for at least 1 minute at 192kb/s 48kHz stereo at ambient temperature

Nagra does not warranty the correct functionality of the machine if other cards than the yellow high lighted ATA cards are used