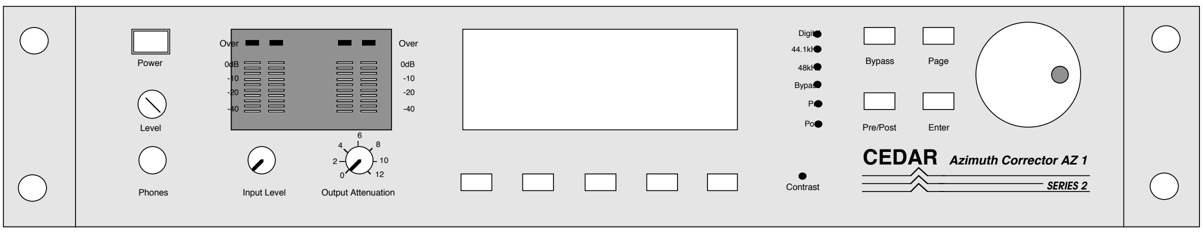
CEDAR

Professional Hardware Systems

AZ-1
Azimuth Corrector

Digital Audio Restoration System

SERIES 2



OWNER'S MANUAL

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INTRODUCTION

Thank you for purchasing the CEDAR AZ-1 Azimuth Corrector. This is the world's most advanced dedicated phase/time correction device, and offers processing power and performance that could only previously be obtained using digital signal processors (DSPs) installed in desk-top (or larger) computer systems such as the CEDAR Production System. The Azimuth Corrector is designed for professional use, although it will work perfectly well in a domestic environment, and its features include the following:

- Real-Time Correction accurate to 1/20 sample.
- Real-Time Detection accurate to 1/4 sample.
- Real-Time Auto-Correction.
- The latest 'SERIES-2' CEDAR hardware.
- Digital Audio interfaces conforming to the AES/EBU and SP-DIF standards.
- 24-bit input and output resolution when using AES/EBU interfaces
- Three sample rates supported on digital inputs: 32kHz, 44.1kHz and 48kHz
- Two sample rates supported on analogue inputs: 44.1kHz and 48kHz
- Balanced analogue inputs and outputs for connection to professional analogue equipment.
- ADC and DAC converters using the latest 64x over-sampling Δ-∑ (Delta-Sigma) technology.
- >103dB dynamic range A/D and >93dB dynamic range D/A
- Mountable in a 19" EIA rack.
- Remote control via MIDI and RS232 interfaces.
- SMPTE/EBU timecode capabilities via optional upgrade
- Input and output LED bar-graph VU meters.
- Twin 40-bit floating point DSP processors delivering 50MFlops to handle the most complex audio processing requirements.
- High levels of artificial intelligence designed into the AZ-1 program algorithms making it extremely simple to use.

THE BACKGROUND TO AZIMUTH CORRECTION

Phase problems and time delays between the left and right channels of a stereo signal account for many of the problems suffered by the audio and video industries. Typical consequences of these errors include poor mono compatibility, poor stereo imaging, loss of high frequencies, and muddy bass response.

Audio technicians often employ a range of processors to hide these deficiencies: equalisers, stereo enhancers, dynamics processors and reverb units. However, none of these attack the heart of the problem - the small but significant non-synchronisation of the left and right channels.

On the basis that, if you remove the cause of the problem the symptoms also disappear, the CEDAR AZ-1 offers timing correction accurate to 1/20th of a sample, enabling you to recover high frequencies and restore imaging that cannot be corrected by other methods. Combining all the facilities of a phase Detector and Corrector, the AZ-1 also features an AUTOTRACKING facility. This enables the unit to measure the delay between the channels and use this value to compensate automatically, in Real-Time, for the delay it detects. This delay is re-calculated 44 times every second, enabling the AZ-1 to compensate for rapidly varying timing errors as well as constant differences.

Since the causes of phase/time errors also often lead to volume imbalances in the stereo signal, the latest and most powerful CEDAR SERIES-2 DSP hardware also features real-time monitoring with gain controls for precise balancing of the corrected signal.

The operation of the AZ-1 is totally digital, and any signal presented to the analogue inputs is converted internally to a suitable digital format by the high quality analogue-to-digital converters (ADCs). Following Time Correction the processed signal is then converted back from digital to analogue by the internal digital-to-analogue converters (DACs).

For use with records, films, video, and tape, no other device offers the power, facilities, or accuracy of the AZ-1.

SAFETY INSTRUCTIONS

CAUTION:

1. **Read all of these instructions**
   All safety and operating instructions should be read before the AZ-1 is operated.

2. **Save these instructions for future reference.**

3. **Follow all warnings and instructions.**

4. **Water and Moisture**
   The AZ-1 should not be used near water, and must not be exposed to rain or moisture. If the AZ-1 is brought directly from a cold environment into a warm one, moisture may condense inside the unit. This, in itself, will not damage the AZ-1, but may cause hazardous electrical shorting to occur. This could severely damage the AZ-1, and even cause danger to life. ALWAYS allow time for the AZ-1 to naturally reach ambient temperatures before connecting the mains power.

5. **Mounting**
   The AZ-1 should be carefully mounted in a 19" EIA rack, or placed on a flat, stable surface. If used on a cart or free stand, care should be taken when moved: uneven surfaces or excessive force may cause cart and AZ-1 to overturn. Do not position the AZ-1 in a place subject to strong sunlight, excessive dust, mechanical vibration or periodic shocks.

6. **Wall or Ceiling Mounting**
   The AZ-1 has not been designed for mounting directly to walls or ceilings.

7. **Ventilation**
   Good air circulation is essential to prevent internal heat built-up within the AZ-1. The AZ-1 should be situated so that its position does not interfere with proper ventilation. The AZ-1 should not be placed in any situation which impedes the flow of air through the vents at the front and rear. Do not place the AZ-1 on a soft surface.

8. **External Heat Sources**
   The AZ-1 should be installed away from significant heat sources such as radiators, and (if possible) away from other audio devices such as amplifiers that produce large amounts of heat. Installation in racks with devices such as signal processors or tape machines should not be a problem.

9. **Power Sources**

   The AZ-1 features an auto-switching power supply which will work safely on any mains supply in the ranges 95v/130v and 190v/260v, 50Hz or 60Hz AC only.

You should never attempt to modify or adjust the internal power supply in any way. It contains no user serviceable parts.

10. **Grounding or Polarisation**
The AZ-1 should always be grounded (or 'earthed').

11. **Power Cord Protection**
Power connectors should be routed so that they will not be walked on or pinched.

12. **Extended Periods of Non-Use**
The AZ-1 is not disconnected from the mains power as long as it is connected to the wall outlet, even if the unit itself has been switched off. Therefore, if the AZ-1 is not to be used for an extended period of time, unplug the unit from the wall. Pull the connector out by the plug, never by the cord itself.

13. **Cleaning**
Clean only with a dry cloth. NEVER use liquid cleaners such as alcohol or benzene on the AZ-1. NEVER use abrasive pads on the AZ-1.

14. **Damage Requiring Service**
The AZ-1 should be returned to qualified service personnel when:
    - objects have fallen into the unit
    - liquid has been spilled into the unit
    - the unit has been exposed to rain
    - the unit fails to function or appears to operate abnormally
    - the unit has been dropped, or the case damaged.

15. **Servicing**
The user should not attempt to service the AZ-1 beyond the instructions contained in the User's Manual. All other servicing should be referred to qualified service personnel.

SET UP

1. UNPACKING AND INSPECTION

Be careful not to damage the AZ-1 during unpacking. Save the carton and all packing materials since you may need them to transport the AZ-1 in the future.

In addition to the packaging, the carton should contain the following:

- mains connection lead
- this manual
- blanking plates which may be used to replace the rack-mount ears

2. INSTALLATION SITE

The AZ-1 may be used in most areas, but to maintain reliability and prolong operating life observe the following environmental considerations:

- Nominal temperature should be maintained between 5˚ and 35˚ Centigrade (41˚ and 95˚ Fahrenheit).

- Relative humidity should be in the range 30% to 60% non-condensing.

- Strong magnetic fields should not exist nearby.

3. RACK MOUNTING

The AZ-1 can be mounted in a standard 19" EIA rack.

4. FREE STANDING USE

The AZ-1 can be used as a free-standing unit. The rack-mount ears may then be replaced by the blanking plates if desired.

To replace the ears with the blanking plates:

- Unscrew the three bolts which attach each ear to the chassis of the AZ-1.

- Attach the blanking plates using the same retaining bolts. Do not over-tighten these bolts as doing so may cause damage to the AZ-1.

CONNECTIONS

The AZ-1 may be connected to most of the professional audio equipment currently available. Three types of audio input and output are provided (one analogue and two digital) and these will satisfy most users' interconnection requirements. Full descriptions of these connectors will be found later in the manual.

1. BEFORE CONNECTION

- To prevent problems and possible equipment damage, turn off the power to all equipment before making connections.

- Be sure to insert plugs firmly into sockets. Loose connections may cause hum and noise.

- When unplugging any lead, do so by grasping the plug, not the lead.

2. POWER CONNECTIONS

Ensure that the AZ-1 is switched OFF before inserting the mains lead.

NOTE: Users with 2-pin mains supplies:
When the AZ-1 is connected to other audio components, the AC hum of the unit may be increased or decreased by reversing the direction of the power connector in the socket. Check that the cord is in the favourable position ('in-phase') with respect to other audio devices in the chain. This will ensure that the best sound quality is obtained from your AZ-1.

For further information on grounding and polarity consult a person familiar with studio grounding techniques.

3. SIGNAL LEAD CONNECTIONS

Refer to the Rear Panel diagram:

The AZ-1 offers three audio connection standards: one analogue and two digital. These are:

- balanced analogue audio I/O
- digital SP-DIF format audio data
- digital AES/EBU format audio data

*Note that the AZ-1 always passes its output to all three signal outputs irrespective of the input used, but that the digital data will only be formatted for **either** AES/EBU **or** SP-DIF, as defined by the user parameters.*

(i) Balanced analogue audio I/O (Pin 2 - 'hot')

This standard is used in professional audio equipment. Connect the output from your source to the balanced analogue inputs of the AZ-1 using standard XLR plugs. You will require two such connections: one for each channel.

The balanced audio output may be used to connect the AZ-1 directly to audio equipment such as mixing desks and professional recorders featuring balanced XLR inputs and outputs.

(ii) Digital SP-DIF format audio data

The SP-DIF format is used by domestic and semi-professional digital audio devices such as DAT machines, some ADCs, and some CD players. Both audio channels are carried along a single cable, so you may connect the SP-DIF output from your source to the SP-DIF input of the AZ-1 using a single cable terminated with RCA (or 'phono') plugs.

The SP-DIF output of the AZ-1 may be connected to the SP-DIF input of your recording device or external DAC.

(iii) Digital AES/EBU format audio data

The digital AES/EBU format is used by professional digital audio devices including mastering systems, DASH recorders, and high quality ADCs & DACs. Both channels of audio are carried along a single cable, so you may connect the AES/EBU output from your source to the AES/EBU input of the AZ-1 using a single cable terminated with XLR plugs.

The AES/EBU output of the AZ-1 may be connected to the AES/EBU input of your digital mixer, recording device or external DAC.

24-bit Digital data resolution:

The AZ-1 features 24-bit input and output resolution whenever the AES/EBU digital input and output are utilised.

Dithering:

The AZ-1 also features TPDF (Triangular Probability Density Function) dithering. This is applied to all data at the 16-bit level unless AES/EBU input is selected. In this case, the data is presented to the digital outputs undithered, but dithering is still applied to the data presented to the DACs.

In order to fully comply with EMC regulations, this unit should be connected via its AES/EBU and/or analogue connectors. Metal-shelled XLR connectors should be used. We recommend using a good quality 'starquad' cable, with *three cores connected to pins 1, 2 & 3. The shield of the cable should be connected, at both ends, to the outer shell of the connector.*

4. OTHER CONNECTIONS

(i) SMPTE/EBU

An optional SMPTE/EBU interface offering LTC and VITC protocols is available for the AZ-1. The standard AZ-1 does not support timecode and these connectors are not present.

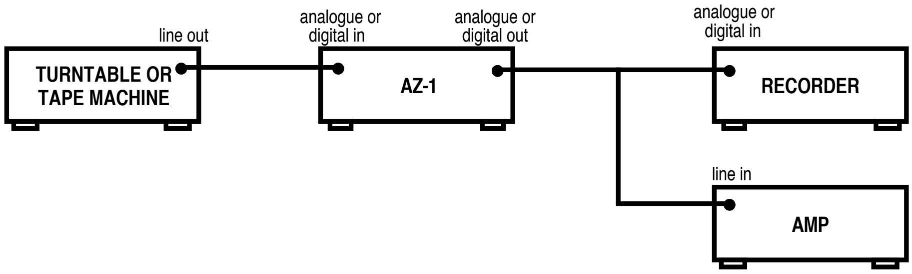
(ii) MIDI IN/OUT/THRU

The operation of the AZ-1 may be controlled using the Musical Instrument Digital Interface (MIDI). Refer to the chapter on Remote Control Protocols for further instructions.

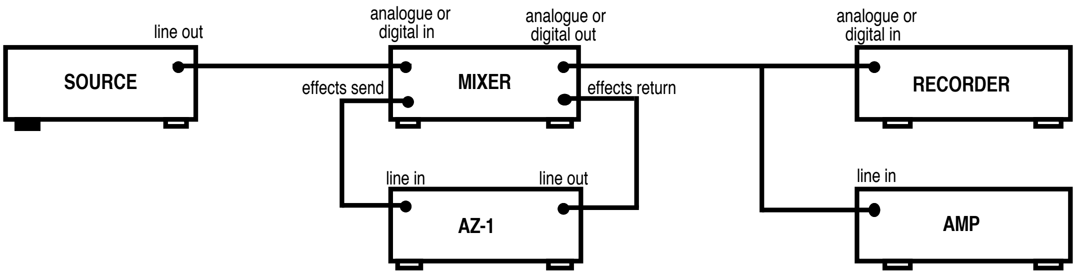
(iii) RS232

The AZ-1 may be controlled using the standard RS232 serial communications protocol. Refer to the chapter on Remote Control Protocols for further instructions.

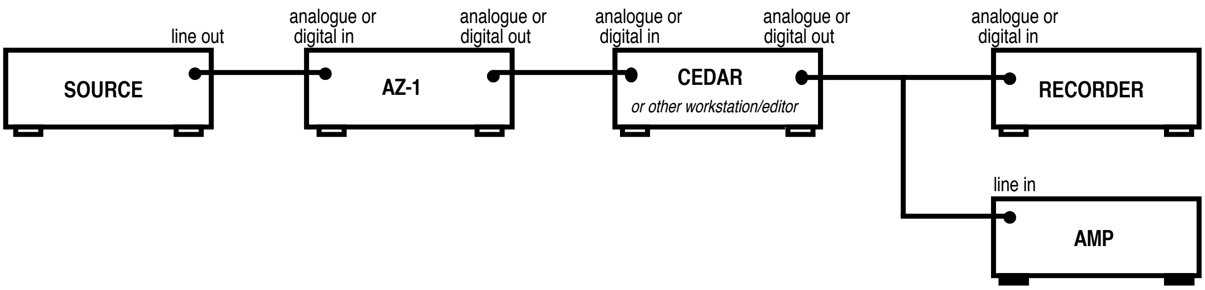
SAMPLE INSTALLATION IDEAS



1. *AZ-1 used in-line for transcription or broadcast purposes.*



2. *AZ-1 used on the effects loop within a studio environment.*



3. *AZ-1 used in-line prior to an editor or audio workstation.*

A GUIDE TO RESTORATION PROCESSING

Contrary to ‘common sense’, the order in which restoration processes are carried out makes a great deal of difference to the quality of the final result. Consequently, there is one ‘right way’ and many ‘wrong ways’ to restore your material.

Following these guidelines will help you to achieve the best results on most material:

- De-Clicking (De-Scratching) should ALWAYS be carried out first. This is because:

  i Large clicks make it difficult for the De-Crackling process to identify and remove the tiny clicks and crackles that constitute surface noise, buzz, and other such problems.

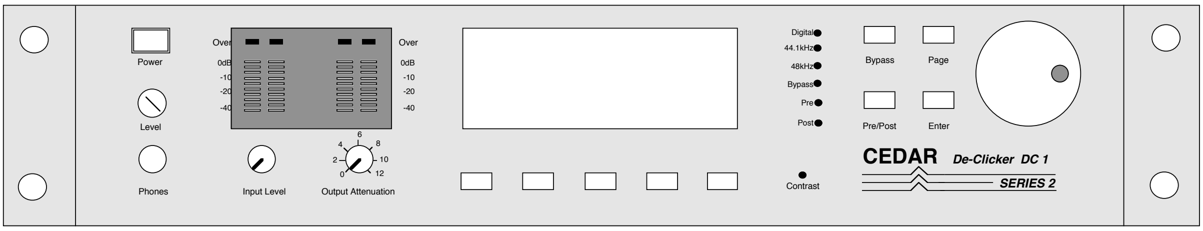
  ii All clicks and scratches are, in effect, tightly defined packets of white noise. If clicks are presented to any of the CEDAR De-Hiss products (HISS-1, HISS-2, DH-1 De-Hiss) they confuse the processes, and create unmusical side effects. In addition, De-Hissing at this stage will make it almost impossible to identify and remove clicks and scratches at a later time.

- De-Crackling should be the next process because even small crackles can cause the same problems as in (ii) above.

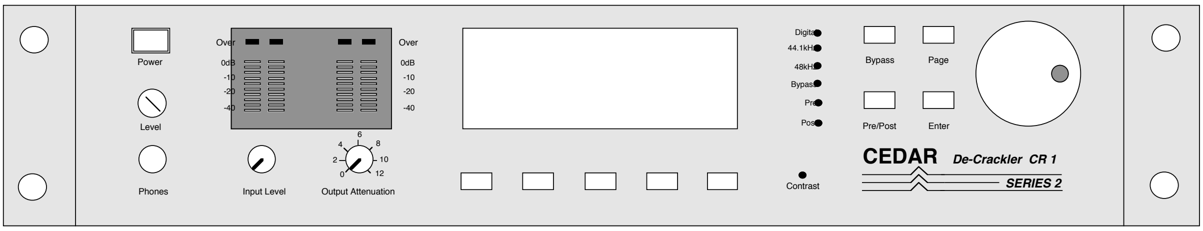
- Azimuth Correction can be carried out either before or after De-Hissing, but experience shows that best results are obtained using the AZ-1 or Phase-EX module before De-Hiss.

- Finally, apply whichever De-Hiss process you wish to use.

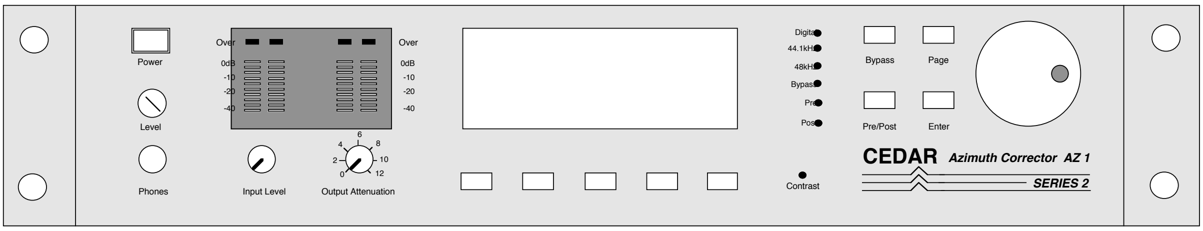
Note: *If you have the full range of CEDAR restoration modules they should be connected as shown in the diagram overleaf. Please note that, to maintain the maximum fidelity and remove and possible sources of degradation between processes, connections between modules should be by AES/EBU (24-bit) format.*



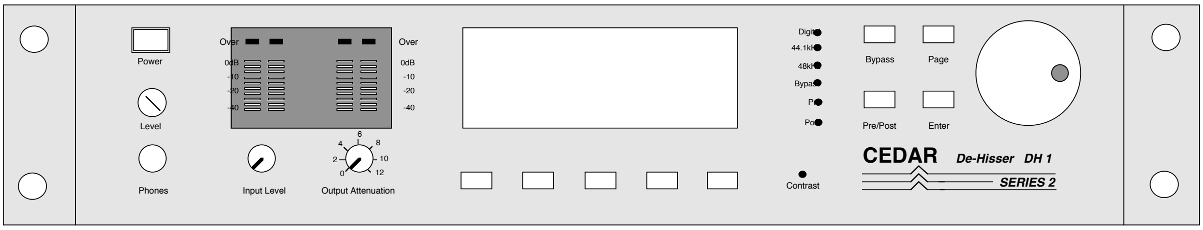
Firstly, De-Click your material



Next, remove crackle and buzz, and reduce distortion if appropriate



Then apply Azimuth Correction to material with phase and balance problems



Finally, apply noise reduction.

LOCATION AND FUNCTION OF FRONT PANEL INDICATORS AND CONTROLS

Refer to the Front Panel diagram:

1. **Power Switch**

2. **Input Signal Meters (Left and Right)**
   Digital signal meters display the peak value of the selected input in dB0s.

   The 'Over' indicators will light if the input signal remains at full scale for four or more consecutive samples.

3. **Output Signal Meters (Left and Right)**
   Calibrated signal meters display the RMS value of all output signals.

   The 'Over' indicators will light if the output signal remains at full scale for four or more consecutive samples.

4. **LCD Screen**
   Provides you with a variety of information and messages, keeping you aware of what is currently happening in the AZ-1.

   All the control screens of the AZ-1 are displayed on the LCD screen. Please refer to the following chapters for full instructions.

5. **Status Indicators**
   Indicate the status of the analogue and digital inputs, and whether the AZ-1 *SERIES 2* is in idle or processing modes.

   Also indicate the possible causes should the unit fail to function.

6. **Dedicated Function Keys.**
   Certain functions are fundamental to operating the AZ-1, and these are controlled by the Dedicated function keys: Bypass, Page, Pre/Post, and Enter.

7. **□-dial (Spinwheel)**
   The □-dial enables you to increase and decrease control values. Please refer to the following chapters for full instructions.

8. **Headphone Socket**
   For use with stereo headphones only. Accepts a standard 1/4" stereo jack plug. DO NOT use 2-conductor mono headphones with the AZ-1.

9. **Headphone Level Control**
   Use this to adjust for a satisfactory listening level. This level control will not alter the signal level at any of the rear panel outputs.

10. **Input Level Control**
This control acts upon the analogue inputs only. Use it to adjust the volume of incoming analogue signals to the desired level. A level of approximately 0 to -3dB (as shown on the Input Signal Meters) will offer best results.

The Input Level Control may be physically bypassed internally to obtain the best possible signal to noise ratio (S/N) from the ADCs. This work must be carried out by qualified service personnel, so please refer to your authorised dealer or directly to CEDAR Audio to have this modification performed.

11. **Output Attenuation Control**
A digital gain control with range 0 to -10dB in 1dB steps.

12. **Function Keys**
Use along with the LCD screen. Please refer to the following chapters for full instructions.

13. **Contrast Control**
The LCD screen may be adjusted for optimum visibility. Use a fine screwdriver to make such adjustments.

QUICK TOUR

If you are impatient to hear some immediate results using your AZ-1 the following instructions should have you up and running within a few minutes:

1. **READ THE SAFETY INSTRUCTIONS.**

2. Connect the AZ-1 to the mains supply.

3. Connect your input and output devices to the AZ-1 using the appropriate input and output sockets. (If in doubt, please refer to the section CONNECTING THE AZ-1 and the manuals of your other equipment).

4. Referring to the front panel diagram, hold down the function key F1 and switch on the AZ-1.

5(i) If you are using analogue inputs press PAGE twice. Press B (function key F2) to select 'analogue'. Then press PAGE twice more to return to the Control Page.

5(ii) If you are using digital inputs from a consumer format machine such as a domestic DAT recorder press PAGE twice, then press B twice to select 'SP-DIF'. If you are outputting to a consumer format machine such as a low-cost DAT recorder press A (function key F1) to select SP-DIF format.

Press PAGE twice to return to the Control Page

Note: The AZ-1 defaults to AES/EBU PROFESSIONAL format, so skip both instructions 4(i) and 4(ii) if your AZ-1 is connected to a system such as the Sony PCM1630.

6. Press B to switch from MANUAL to AUTO.

7. Play your material.

8. Press PRE/POST to hear an immediate difference between the processed and unprocessed signals (assuming, of course, that your original material suffers from some form of azimuth or other timing error).

This section should have whetted your appetite, so you should now proceed to the rest of the manual to ensure that you can obtain the best results from your CEDAR AZ-1.

WARMSTART AND COLDSTART

The AZ-1 features Warmstart and Coldstart options. Warmstart has been added so that the unit can be configured once, and these parameters are then automatically recalled on every power-up. This is ideal for applications where time-consuming set-ups at the start of each session are not practical.

Coldstart

If the AZ-1 has not been used for some time the system will automatically Coldstart. This process initialises all parameters to their factory default values, and after a few seconds the AZ-1 will automatically enter at Page 1.

On start-up the message 'Coldstart' will be displayed at the top right of the start-up screen on the LCD display. The screen will then enter PAGE 1, which will show the default Parameters:

The default values are:	Time	=	0
	Detect Mode	=	Manual
	Output Mode	=	Stereo
	Phase	=	L+ R+
	Balance	=	0
Other default values are:	Digital Output	=	AES/EBU
	Input Source	=	AES/EBU
	Receiver Error Level	=	1 - Lock
	MIDI	=	Channel 1
	Bypass	=	OFF
	A to D frequency	=	44.1kHz
	Pre/Post	=	Post

Warmstart

The AZ-1 remembers the latest parameters used, and the page that was active at the time that the system was last switched off.

On start-up the AZ-1 will display the message 'Warmstart' on the screen, and after a few seconds will re-enter at the appropriate page, with all user parameters set to their previous values.

User Coldstart

If you wish to force the AZ-1 to Coldstart, hold down Function Key F1 while switching on the system. Release F1 when the message Coldstart is seen on the LCD display.

*Note: In common with all other digital devices, and irrespective of whether you are Warmstarting or Coldstarting the AZ-1, you should always allow a few seconds between switching the unit **off**, and switching it **on** again.*

OPERATING THE CEDAR AZ-1

1. DEDICATED CONTROLS:

The AZ-1 features a number of dedicated controls to speed operation. These are:

Dedicated Function Keys:

- Bypass
- Page
- Pre/Post
- Enter

I/O Level Controls

- Input Level
- Output Attenuation

These are now explained in turn:

Bypass

You may wish to bypass completely the operation of the AZ-1. Press BYPASS to do this. The current status will be indicated on the Status LED.

The Bypass does not 'hard-wire' the input to the output. Analogue signals still pass through the AtoD and DtoA stages.

Notes:

- *There is a delay of approximately 1.3mS in any analogue-to-analogue signal passed through the AZ-1 in Bypass mode.*
- *There is a delay of approximately 0.1mS in any digital-to-digital signal passed through the AZ-1 in Bypass mode.*
- *All delays are 'group delays' (i.e. are constant at all frequencies) and are measured at a sample rate of 44.1kHz.*

Page

Use this Function Key to move between Pages.

Pre/Post

It will often be useful to compare the original signal with the post-processing output of the AZ-1. The current status will be indicated on the Status LEDs.

When set to 'PRE', the Pre/Post control bypasses the PHASE CORRECTOR and CHANNEL GAINS (see page 19).

Enter

The ENTER Key has three functions: as a LOCK-OUT key, preventing accidental changing of parameters; as a CLEAR key, resetting error messages, and as a MIDI DUMP command.

These first two functions are, of course, context sensitive, and the key's action will be appropriate to the page displayed (see below). The MIDI DUMP will be initiated every time that the ENTER key is pressed, regardless of context.

Input Level

This control acts upon the analogue inputs only. Use it to adjust the volume of incoming signals to the desired level. We recommend a peak level of approximately 0 to -3dB as shown on the Input Signal Meters.

Output Attenuation

Avoid clipping using the Output Attenuation Control. This is not a compressor or limiter, and acts purely as a digital gain control with variable gain from 0dB to -10dB in 1dB steps.

OPERATING THE CEDAR AZ-1

2. PAGES:

The AZ-1 has four 'pages' which control all aspects of its operation. Each page is displayed on the LCD screen, and may be controlled using the Function Keys and the □-dial.

Switch the AZ-1 on. (Refer first to the safety instructions.)

The screen will immediately enter the CONTROL PAGE, which will show the Warmstart parameters stored when the unit was last used.

All the controls for the AZ-1 are contained in three of the four PAGES, each of which is selected by pressing the dedicated **PAGE** function key. The Pages are cycled, and will appear in the following order:

- Control Page
- XY Display Page
- I/O Control Page
- Remote Control Page

These, and a further description of the Dedicated Controls, are now covered in turn.

Note: There is a fifth, normally hidden, page called the Status Page. This is not accessed using the standard 'Page' function, and will be discussed separately in the section describing Error Levels.

PAGE 1: CONTROL PAGE

If necessary, access this page by pressing the Dedicated Function Key PAGE until the Control Page appears.

There are five controls in the Control page. These correspond to the five 'soft-keys' and are to be found directly above each of them as follows:

F1 • Time Control
F2 • Detect Mode Control
F3 • Output Mode Control
F4 • Phase Mode Control
F5 • Balance Control

The controls, and therefore the AZ-1 itself, act in the following order:

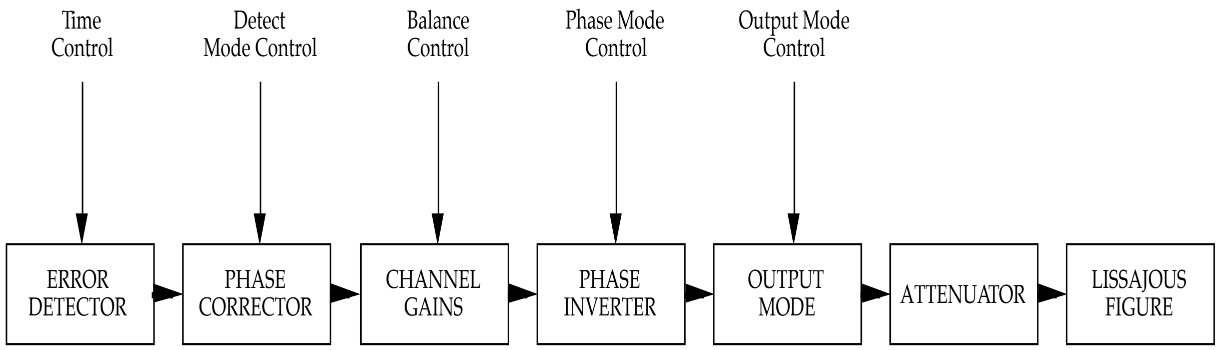


Figure 1: The order in which the CONTROL PAGE controls modify the signal.

You can optimise the beneficial effect of the AZ-1 by setting each of these controls appropriately. They are now described in turn:

Detect Mode:

The AZ-1 may be used in two ways: in MANUAL mode as a manual time-shifter, or in AUTO mode as an AUTOTRACKING system, in which the AZ-1 determines the time error between channels and automatically corrects this.

Press F2 to toggle between modes.

Time:

TIME refers to the group delay (azimuth error) existing between the left and right channels of your signal. It is displayed in both numerical and graphical formats.

In PRE mode:

When you are monitoring your signal in PRE mode, TIME simply displays the error.

In POST mode: (i) with MANUAL selected:

When you are monitoring your signal in POST mode, TIME displays the amount of correction being applied to the signal.

TIME may be manually altered from +99.00 to -99.00. A change of 1.00 corresponds exactly to a shift of 1 digital word.

TIME may be manually adjusted as follows:

- If the TIME control is **not** already highlighted, press F1 to select it. A box will appear around the numerical display to indicate that the control is selected.
- TIME may be altered in steps as fine as 0.01 samples (i.e. 1% of a sample, or approximately 1/5,000,000th of a second at 48kHz sampling rate). Rotate the □-dial clockwise and/or anti-clockwise to alter TIME.
- The action of the □-dial is velocity sensitive, and quick rotation will result in a larger change of values than will slow rotation.

To reset TIME to 0.00 press the the F1 key once more whilst TIME is highlighted.

In POST mode: (ii) with AUTO selected:

With AUTO selected in POST mode, the AZ-1 measures the amount of azimuth error and automatically applies a compensating correction to the signal. The TIME display shows the amount of correction applied.

It is not possible to manually alter the amount of correction in AUTO mode.

Note: A TIME shift of 1.00 is equivalent to using an editor to record your signal; remove or insert a single sample at the start of either the left or right channels; and then play back the audio file.

Balance:

Any electronic problems leading to phase/time errors will often cause channel imbalances (i.e. unmatched channel gains). BALANCE allows you to balance the left and right channels of your signal.

The relationship between the BALANCE and the GAIN applied to each channel is shown in the following graph. This shows the gain in dB applied to each channel as you sweep the BALANCE from 0.00 to +99.00 or -99.00.

The BALANCE setting may be manually adjusted as follows:

- If the BALANCE control is **not** already highlighted, press F5 to select it. A box will appear around the numerical display to indicate that the control is selected.
- BALANCE may be altered in steps as fine as 0.01. Rotate the □-dial clockwise and/or anti-clockwise to alter BALANCE.
- The action of the □-dial is velocity sensitive, and quick rotation will result in a larger change of values than will slow rotation.

To reset BALANCE to 0.00 press the the F5 key once more whilst BALANCE is highlighted.

Phase Mode:

The AZ-1 allows you to invert the phase of the left and right channels, either individually or together. Normal operation is indicated by L+R+, but you may also select:

L- R+	the left channel only is inverted (180° phase shift)
L- R-	both channels are inverted (no relative shift between channels)
L+ R-	the right channel only is inverted (180° phase shift)

Cycle between phase modes by pressing F4 until the desired mode is selected.

Output Mode:

Three output modes for monitoring the final signal are provided:

- Stereo

  The left and right channels of the input signal are analysed and corrected (or not, depending upon other settings) and are then passed to the left and right outputs.

- Mono +

  The left and right channels of the input signal are analysed and corrected (or not, depending upon other settings) and are then summed together. The summed mono signal is then passed to the left and right outputs.

- Mono -

  The left and right channels of the input signal are analysed and corrected (or not, depending upon other settings) and the left channel is then subtracted from the right channel. The subtracted mono signal is then passed to the left and right outputs.

  Note: If the input is perfect mono then (because the channels perfectly cancel each other out, and provided that the AZ-1 is not used to alter one of the other of the channels) the output in Mono - will be silence.

Cycle between output modes by pressing F3 until the desired mode is selected.

PAGE 2: XY DISPLAY

Access this page by repeatedly pressing the Dedicated Function Key PAGE until the XY DISPLAY appears.

This page shows the Lissajous figure in large format. The following displays may be seen, depending upon the type of material offered to the AZ-1:

Figure 3(a)	Pure in-phase mono signal.
Figure 3(b)	Pure out-of-phase mono signal.
Figure 3(c)	Left channel only.
Figure 3(d)	Right channel only.
Figure 3(e)	Stereo signal with poor phase correlation.
Figure 3(f)	Stereo signal with good phase correlation.

When using the AZ-1 as an azimuth corrector for a stereo signal, you should notice that, if in PRE mode the signal resembles 3(e), toggling to POST changes the figure to look more like 3(f). This demonstrates that the correlation between channels is improved by the AZ-1.

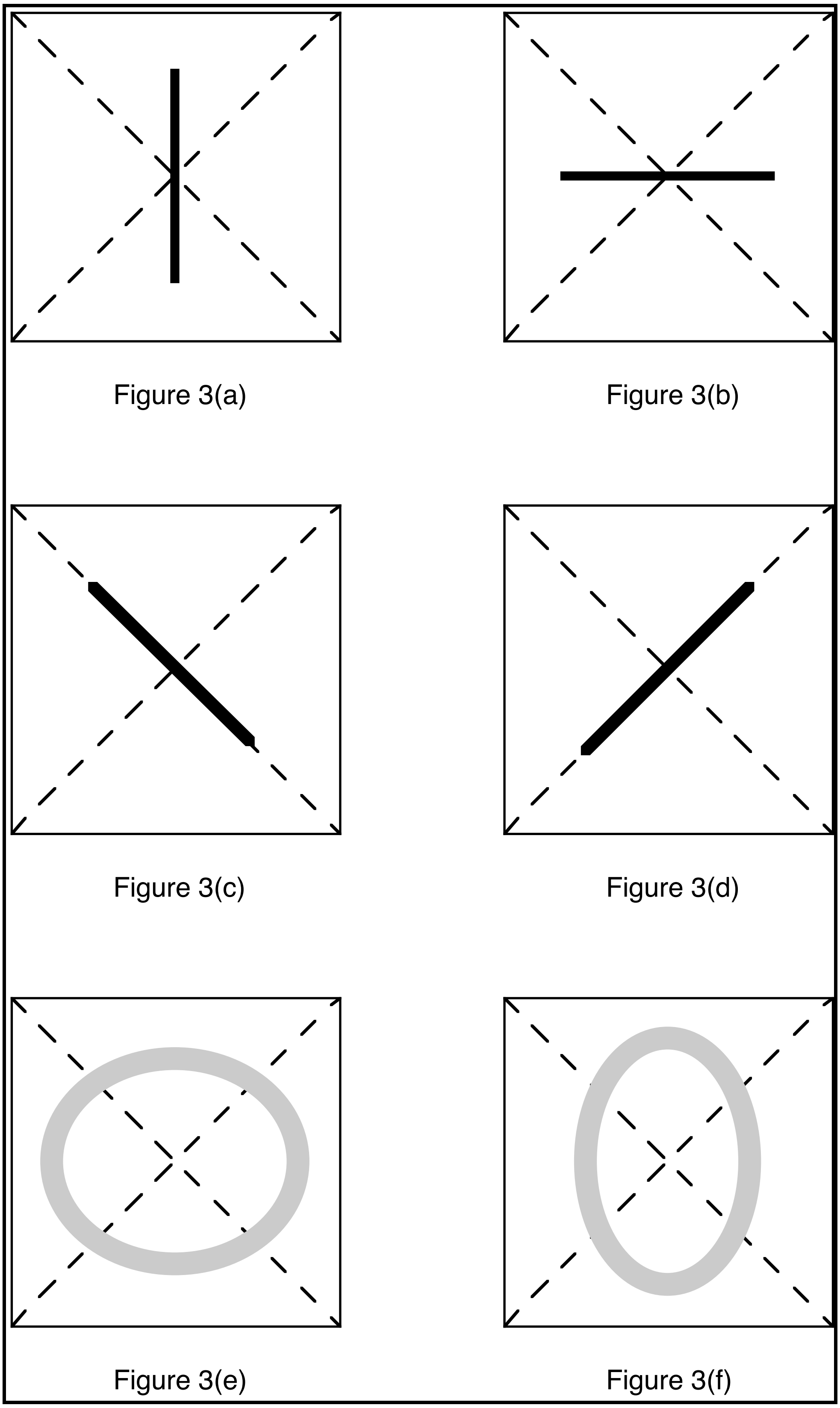


Figure 3: Examples of Lissajous figures encountered in audio signals.

PAGE 3: INPUT/OUTPUT CONTROL PAGE (I/O CONTROL)

Access this page by repeatedly pressing the Dedicated Function Key PAGE until the I/O CONTROL PAGE appears.

This page allows you to determine the input used, the sampling frequency of the Analogue to Digital Converters, the digital input error detection level, and the digital output format.

(Remember that all outputs are permanently active, and that they do not require selecting, but that the same digital data is supplied to both AES/EBU and SP-DIF outputs. The data format will therefore only be appropriate for one digital output at any given time.)

There are three options in the I/O Control Page:

A. DIGITAL OUTPUT:

This option defaults to AES/EBU. To toggle between the two output modes, AES/EBU and SP-DIF, press the Function Key marked 'A' on the LCD screen.

- AES/EBU FORMAT:

  When AES/EBU is selected, both the phono and XLR connectors will carry AES/EBU specification audio data. You should patch the output from the XLR connectors to your recording device.

  The AZ-1 features 24-bit input and output resolution when AES/EBU is selected.

- SP-DIF FORMAT:

  When SP-DIF is selected, both the phono and XLR connectors will carry SP-DIF specification audio data. You should patch the output from the phono connectors to your recording device.

  TPDF dithering will be applied to the digital data at the 16-bit level.

B. INPUT SOURCE:

There are three input sources: AES/EBU, SP-DIF and ANALOGUE.

To toggle between the input sources press the Function Key marked 'B' on the LCD screen. The Status LEDs will indicate the inputs selected and the sample rate received (digital) or selected for conversion (analogue).

- SAMPLE RATE OF INCOMING DIGITAL SIGNAL:

  When the AZ-1 is switched to receive digital audio data, the 'DIGITAL' LED will be lit, and the front panel LEDs will indicate the sample rate of the digital signal presented to the inputs:

neither 44.1 nor 48 kHz LED lit	=	32 kHz signal presented to inputs
44.1 kHz LED lit	=	44.1 kHz signal presented to inputs
48 kHz LED lit	=	48 kHz signal presented to inputs

- CLOCK DETECTION:

  If the AZ-1 fails to detect a digital signal within the following limits, the 44.1kHz and 48kHz LEDs will flash continually. This will be irrespective of any other system settings.

Acceptable ranges:	44.1kHz	±	4%
	48kHz	±	4%
	32kHz	±	4%

- SAMPLE RATE OF A TO D CONVERTERS

  When the AZ-1 is switched to receive analogue audio data, the 'DIGITAL' LED will not be lit, and the front panel LEDs will indicate the sample rate of the analogue-to-digital converters.

  The ADCs in the AZ-1 do not offer a 32kHz option unless synchronised to an external 32kHz source.

C. A TO D FREQUENCY (INPUT SOURCE = ANALOGUE)

The ADC frequency may be selected by two, fundamentally different, methods. The first is to select one of the internal clock frequencies available, the second is to control the sample rate by using an external clock.

- INTERNAL CLOCK FREQUENCIES

  To toggle between the AZ-1s internal 44.1kHz and 48kHz sampling frequencies (and between AES Sync and SP-DIF Sync - see below) press the Function Key marked 'C' on the LCD screen. The change in frequency will be shown on-screen and also by the Status LEDs.

  Note: The sampling frequency reverts to 44.1kHz on Coldstart.

- EXTERNAL SYNCHRONISATION

  The AZ-1 clock may be synchronised to either the AES/EBU input or the SP-DIF input. Connecting a valid digital input to either of these and selecting AES Sync or SP-DIF Sync (as appropriate) will lock the AZ-1 to the external clock.

  If the external clock falls within the acceptable ranges of each of the standard sample rates (44.1kHz, 48kHz, 32kHz) the clock frequency will be shown on the LEDs. If the external clock lies outside these ranges the AZ-1 will still function, and good audio will be produced at the analogue output. Whether the digital output will be usable will then be determined by the flexibility of other devices in the digital audio chain.

  To toggle between AES Sync and SP-DIF Sync (and also between the AZ-1s internal 44.1kHz and 48kHz sampling frequencies) press the Function Key marked 'C' on the LCD screen.

  Note: If external synchronisation is requested, but no valid signal is detected at the appropriate digital input, the DIGITAL LED will flash to indicate the error.

D. RECEIVER ERROR LEVEL (INPUT SOURCE = AES/EBU or SP-DIF)

The AZ-1 features sophisticated software which detects and analyses both fatal and non-fatal errors in the incoming digital audio data.

You may select one of four error levels which will cause the front panel 'DIGITAL' LED to flash if the incoming data contains an error equal to or worse than the selected level.

The error levels are:

- **1 - Lock**

  This is the 'weakest' detector and will only cause the LED to flash when the AZ-1 believes that there is no usable signal being presented to the selected digital input.

- **2 - Code**

  If there is an incoming signal yet the LED flashes on error level 2, the AZ-1 is indicating that the signal contains coding violations. In some cases you may obtain usable audio. However, this warning may be caused by non-AES/EBU or non-SP-DIF data being presented. In these cases any audio produced will almost certainly be unusable.

- **3 - Trans**

  This indicates that the incoming digital audio data is of poor quality (i.e very noisy or jittery) and that undetectable data errors are likely. These errors will not be corrected by any standard AES/EBU or SP-DIF device and may lead to audio degradations.

- **4 - Valid**

  This is the most stringent test of the incoming data, and will cause the LED to flash if the AZ-1 determines that any of the data contained in the signal is not valid. This is often non-fatal (i.e. you will hear perfectly good audio) but it indicates that some device or anomaly in your audio chain is generating digital audio data outside of the AES/EBU or SP-DIF specifications published by their respective bodies. Please note however that, if the digital LED does not flash, this can not be taken as an absolute statement that the signal conforms to specification.

Note: If the error level selected detects an error, the digital audio signal will be coded as INVALID by the AZ-1. Many manufacturers' devices do not recognise or act upon this code, but those that do may refuse to accept or record the audio.

PAGE 4: REMOTE CONTROL

Access this page by repeatedly pressing the Dedicated Function Key PAGE until the REMOTE CONTROL PAGE appears.

The AZ-1 features intelligent 'auto-detection' software which monitors the RS232, MIDI, and SMPTE/EBU (if fitted) inputs and responds to data received on each and any of them. This eliminates the need for a control to select the remote control to be used.

It is only necessary, therefore, to select the Channel on which the AZ-1 receives commands over MIDI.

MIDI

CEDAR Audio Ltd do not produce software for remote devices to control the AZ-1 over MIDI.

- MIDI CHANNEL

  Ensure that button A is highlighted by a box. It is then possible to change the MIDI Channel turn the □-dial clockwise (to increase) or anti-clockwise (to decrease) the MIDI Channel.

  To toggle this function on/off press the Function Key marked 'A'.
  On Coldstart the MIDI Channel defaults to 1.

RS232

CEDAR Audio Ltd do not produce software for remote devices to control the AZ-1 over RS232. However, for users wishing to implement their own control software, the RS232 Protocol is outlined in the chapter 'RS232 Protocol'.

SMPTE/EBU Timecode

A separate SMPTE/EBU reader/generator board may be purchased and fitted inside your AZ-1. Please contact your dealer for details of this.

PAGE 5: STATUS PAGE

Access the STATUS PAGE by holding down Function Key F5 and then pressing the Dedicated Function Key PAGE.

Should the AZ-1 fail to function, or appear to function incorrectly, there may be an error contained within the digital audio data received at the System's inputs. The Receiver Error Level (see above) will notify you when an error has occurred, but it will not tell you what it is. For many users, this information will be adequate, but the AZ-1 is capable of reporting errors and other status information in more detail.

The STATUS PAGE will give you information regarding the current status of the AZ-1, and will give you details regarding any errors which have occurred since the unit was switched on.

Three items of information will always be reported by the AZ-1. These are:

- DSP1: Status Crashed / Timed Out / Running
- DSP2: Status Crashed / Timed Out / Running
- I/O: Condition Error / Emphasis, Sample Rate

If a remote control error is detected, a fourth field will appear:

- Comms: Error Illegal Checkbyte / Illegal Command Size

STATUS INDICATORS

The front panel LEDs will help to identify the possible cause if the unit fails to function. The following table lists all possible combinations of LED error indications:

LED flashing:		**Condition:**
Digital	-	the digital input violates the Receiver Error Level
	-	*or* no digital sync is present (if requested in I/O page)
44.1 *and* 48kHz	-	unknown sample rate received at inputs
Bypass/Pre/Post	-	One or both of the DSPs have crashed.

STATUS PAGE DEFINITIONS:

Crashed

The AZ-1 DSPs are failing to function. The only recourse is to switch the unit off, wait for a few seconds, and then switch on again. If this error re-occurs please refer your AZ-1 to an authorised service centre.

Timed Out

If, for any reason, the AZ-1 drops out of real-time (fails to pass audio to the output) this error will be reported. This should only occur if a sample rate of greater than 50kHz is presented to one of the digital inputs. This error is non-fatal, and the AZ-1 should continue to function normally after it has occurred.

Running

The AZ-1 DSPs are functioning correctly and, moreover, have been doing so since the unit was switched on.

Error

If the DIGITAL LED is flashing the most serious error will be detailed at this point. Errors are fully detailed in the AZ-1 Service Manual.

Emphasis

If no error is detected, the I/O status will display the Emphasis condition:

- OFF

The Emphasis bit is not set.
The DAC de-emphasis will not be engaged.

- 50/15

The Emphasis bit is set to 50/15 µS.
The DAC de-emphasis will be engaged.

- J17 (AES/EBU only)

The Emphasis bit is set to CCITT J17.
The DAC de-emphasis will not be engaged.

- Unknown (AES/EBU only)

The Emphasis status is not indicated.
The DAC emphasis status will not be altered.

Sample Rate

If no digital data error is detected, the measured sample rate presented to the digital inputs will be displayed to the nearest 100Hz.

Illegal Checkbyte

The RS232 or MIDI has received a command packet containing an illegal checkbyte (byte2).

Illegal Command Type	The RS232 or MIDI has received a command packet containing an illegal command type (byte4).

TUTORIAL

Assuming that everything is connected correctly, you will be able to leave the AZ-1 permanently in your audio signal path without risking any damage or degradation to the signal.

If MANUAL mode is selected and TIME is set to 0.00, or if the unit is in PRE or BYPASS modes, the AZ-1 will act purely as a sophisticated phase meter. If you then leave the AZ-1 in its Control or Display Pages it will soon be apparent if you have Azimuth errors in any material passed through it.

If you detect azimuth errors and decide that they require correction the following methodology may be of assistance:

1. Ensure that the AZ-1 is in POST and that BYPASS is OFF.

2. The best results will be obtained if the audio is left-right balanced, so your first job will be to select the Control Page and adjust the BALANCE control to balance the left and right channels.

3. To get a good idea of the time difference between channels, put the AZ-1 into MANUAL and MONO- (mono difference) OUTPUT MODE.

4. You can now sweep the time relationship between channels using the TIME control. Do so until the perceived signal is minimised. *HINT: The TIME value will usually lie between +10 and -10, so don't be surprised if you never seem to need larger values.* (Often you can recognise this by noticing that vocalists and solo instruments disappear from the mix, leaving just the reverberation behind.)

   You will also be able to fine tune the BALANCE in MONO-. Remember that the best results will always be obtained when both BALANCE and TIME are correctly set for each piece of material.

5. Return to STEREO mode, switch the AZ-1 to AUTO, and use PRE/POST to compare the uncorrected and corrected signals. Unless the errors in the signal are swinging wildly, the typical time-shift in AUTO should approximate to the MANUAL time-shift you detected.

6. If satisfied with the results, re-start the audio material and allow the AZ-1 to process as it decides appropriate. Monitor the XY and TIME DISPLAYS to ensure that the signal is not 'confusing' the AZ-1. If these displays swing dramatically you may be processing audio which contains unusual stereo information, or modern audio effects such as ADT which can make it difficult for the AZ-1 to detect and correct the azimuth error. In these cases, it will be necessary to use MANUAL mode to process the audio.

7. Having decided whether to use AUTO or MANUAL play the material through the AZ-1.

Note: We would be grateful to receive feedback from our users regarding their experiences with the AZ-1. Any suitable hints and tips will be included in this tutorial in later versions of the manual.

REMOTE CONTROL PROTOCOLS

1. RS232

RS232 is defined in the AZ-1 *SERIES 2* as:

9600 baud
8 bits data
1 stop bit
No parity

A command packet contains 6 bytes. These are:

byte 1:	channel number byte: must be 0xAF
byte 2:	Checkbyte. Fixed: must be 0x63
byte 3:	command number (see below)
byte 4:	Command type. Fixed: 0x07
byte 5:	command value HIGH byte
byte 6:	command value LOW byte

The HIGH and LOW bytes together form a signed integer.

Command Numbers:		**Command Values:**		
0xF7	Clear Errors command	Any value	=	Clear all error messages
0xF8	Select Page command	1	=	Control Page
		6	=	I/O Control Page
		7	=	Status Page
		15	=	Remote Control Page
		-1	=	Toggle between Pages
		Any other value	=	Refresh
0xF9	Pre/Post command	0	=	Pre
		1	=	Post
		-1	=	Toggle
		Any other value	=	Refresh
0xFA	Bypass command	0	=	Bypass OFF
		1	=	Bypass ON
		2	=	RESERVED VALUE
		3	=	RESERVED VALUE
		-1	=	Toggle
		Any other value	=	Refresh
0xC0	Digital Output Format	0x80	=	SP-DIF
		0x00	=	AES/EBU
		-1	=	Toggle
		Any other value	=	Refresh

0xC1	Input Source	0	=	Analogue
		1	=	SP-DIF
		2	=	AES/EBU
		-1	=	Toggle
		Any other value	=	Refresh
0xC2	A to D Frequency	0	=	44.1kHz
		1	=	48kHz
		2	=	SP-DIF Sync
		3	=	AES/EBU Sync
		-1	=	Toggle
		Any other value	=	Refresh
0xC3	Receiver Error Level	0	=	1 - Lock
		1	=	2 - Code
		2	=	3 - Trans
		3	=	4 - Valid
		-1	=	Toggle
		Any other value	=	Refresh
0x22	Detect Mode	0	=	Manual
		1	=	Auto
		-1	=	Toggle
		Any other value	=	Refresh
0x23	Output Mode	0	=	Stereo
		1	=	Mono +
		2	=	Mono -
		-1	=	Toggle
		Any other value	=	Refresh
0x24	Phase Mode	0	=	L+R+
		1	=	L-R+
		2	=	L-R-
		3	=	L+R-
		-1	=	Toggle
		Any other value	=	Refresh
0x20	Set TIME	Any value	=	TIME x 100
	This command will be ignored in AUTO mode.			
0x30	Alter TIME	Any value	=	∆ (TIME) x 100
	This command will be ignored in AUTO mode.			
0x21	Set BALANCE	Any value	=	BALANCE x 100
0x31	Alter BALANCE	Any vlaue	=	∆ (BALANCE) x 100

2. MIDI

The AZ-1 is permanently set to transmit any change of control page parameters or Pre/Post state via MIDI except when such a change is initiated by an RS232 or MIDI command. Therefore, if a MIDI sequencer such as Cubase™, Notator™, or EditTrack™ is connected to the AZ-1, it will receive a running history of the unit's operation.

If your sequencer and audio sources are able to send and receive timecode, then the AZ-1's MIDI capability may be used as the basis for an automation system.

Note: The absolute parameter values are not transmitted or received, so the user must ensure that any changes are relative to a desired starting value which can be set using MIDI DUMP.

If a MIDI DUMP of all control page parameters and the Pre/Post state is required, pressing ENTER at any time will initiate the DUMP.

Additional MIDI Command

The AZ-1 will receive LOCAL ON and LOCAL OFF commands.
The Status Page will notify you of the current state.
Both WARMSTART and COLDSTART always set LOCAL ON.

This command cannot be initiated from the front panel of the AZ-1.

SELF TEST MODE

The AZ-1 *SERIES 2* features a powerful self-test mode which enables the System to check the operation of each of its major sub-systems, plus all of the user controls.

To enter the self-test mode:
Switch on the AZ-1 *SERIES 2* while holding down the ENTER key. The AZ-1 will perform each test in turn, and you may move to the next test by pressing the ENTER key. Consequently, any test may be skipped by pressing the ENTER key.

Note: Whilst the SELF-TEST is in progress, the ENTER key will not initiate a MIDI DUMP.

ROUTINE 1: BUTTON TESTING ROUTINE

The AZ-1 *SERIES 2* will invite you to press each of the Function Keys (except ENTER) and each of the Dedicated Function Keys. Pressing a key will cause the display to change from OFF to ON.

ROUTINE 2: ATTENUATION KNOB TEST

The AZ-1 *SERIES 2* will invite you to turn the Attenuation knob to check that the value displayed on screen matches the position of the knob..

ROUTINE 3: □-dial (SPIN WHEEL) TEST

Rotate the □-dial to check that values change smoothly in both positive (clockwise) and negative (anti-clockwise) directions.

ROUTINE 4: LED TEST

The AZ-1 *SERIES 2* will flash all six green LEDs.

ROUTINE 5: METER TEST

The AZ-1 *SERIES 2* will invite you to turn the □-dial to vary the levels displayed by each of the four input and output meters in turn. Press ENTER to step to the next meter.

ROUTINE 6: DSP1 TEST

The AZ-1 *SERIES 2* will test its DSPs and internal memory. Please wait for this test to complete.

- If the System is fully functional the screen will display the message: **"Memory passed".**

- If a memory error is detected the screen will display the message: **"Memory error at: .....".**

- If a DSP failure is detected the screen will display the message: **"DSP1 is not responding".**

If you observe this message please repeat the self-test. If the message recurs please contact your dealer for assistance.

WARNING: *The AZ-1 SERIES 2 contains no user-serviceable parts. DO NOT UNDER ANY CIRCUMSTANCES attempt to service your unit.*

ROUTINE 7: DSP2 TEST

As above.

TEST COMPETED

Your AZ-1 *SERIES 2* will now prompt you to press ENTER one more time to return you to operating mode (whether all tests have been passed or not).

Some failures will not stop you from using the AZ-1 *SERIES 2* successfully. However, consistent failures should be notified to your dealer or directly to CEDAR Audio Ltd.

CEDAR AZ-1

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