CEDAR

BRX+ debuzzer
AZX+ azimuth corrector

Digital Audio Restoration Modules

OWNER'S MANUAL

Manual version 1.00: February 1999

LICENCE AND LIMITED WARRANTY

1. DEFINITIONS

In this Licence and Limited Warranty the following words and phrases shall bear the following meanings:

'the Company'	means CEDAR Audio Limited of 9 Clifton Court, Cambridge CB1 7BN, UK;
'the System'	means an instance of the sound-reprocessing system comprising hardware and software held on EPROM ('firmware') developed by the Company;
'this Document'	means this Licence and Limited Warranty.

2. ISSUE AND USE OF THE SYSTEM

2.1 The terms and conditions of this Document are implicitly accepted by any person or body corporate who shall at any time use or have access to the System, and are effective from the date of supply of the System by CEDAR Audio Limited to its immediate customer.

2.2 The Company hereby grants to the Licensee and the Licensee agrees to accept a non-exclusive right to use the System.

3. PROPERTY AND CONFIDENTIALITY

3.1 The System contains confidential information of the Company and all copyright, trade marks, trade names, styles and logos and other intellectual property rights in the System including all documentation and manuals relating thereto are the exclusive property of the Company. The Licensee acknowledges that all such rights are the property of the Company and shall not question or dispute the ownership of any such rights nor use or adopt any trading name or style similar to that of the Company.

3.2 The Licensee shall not attempt to reverse engineer, modify, copy, merge or transcribe the whole or any part of the System or any information or documentation relating thereto.

3.3 The Licensee shall take all reasonable steps to protect the confidential information and intellectual property rights of the Company.

4. LIMITED WARRANTY AND POST-PARTWARRANTY OBLIGATIONS

4.1 The Company warrants that the System will perform substantially in accordance with the appropriate section of its accompanying product manual for a period of one year from the date of supply to the Company's immediate customers.

4.2 The Company will make good at its own expenses by repair or replacement any defect or failure that develops in the System within one year of supply to the Company's immediate customer.

4.3 The Company shall have no liability to remedy any defect, failure, error or malfunction that arises as a result of any improper use, operation or neglect of the System, or any attempt to repair or modify the System by any person other than the Company or a person appointed with the Company's prior written consent.

4.4 In the case of any defect or failure in the System occurring more than twelve months after its supply to the Company's immediate customer the Company will at its option and for a reasonable fee make good such defect or failure by repair or replacement (at the option of the Company) subject to the faulty equipment having first been returned to the Company. The Company will use reasonable efforts to return repaired or replacement items promptly, all shipping, handling and insurance costs being for the account of the Licensee.

4.5 The above undertakings 4.1 to 4.4 are accepted by the Licensee in lieu of any other legal remedy in respect of any defect or failure occurring during the said period and of any other obligations or warranties expressed or implied including but not limited to the implied warranties of saleability and fitness for a specific purpose.

4.6 The Licensee hereby acknowledges and accepts that nothing in this Document shall impose upon the Company any obligation to repair or replace any item after a time when it is no longer produced or offered for supply by the Company or which the Company certifies has been superseded by a later version or has become obsolete.

5. FORCE MAJEURE

The Company shall not be liable for any breach of its obligations hereunder resulting from causes beyond its reasonable control including, but not limited to, fires, strikes (of its own or other employees), insurrection or riots, embargoes, container shortages, wrecks or delays in transportation, inability to obtain supplies and raw materials, or requirements or regulations of any civil or military authority.

6. WAIVER

The waiver by either party of a breach of the provisions hereof by the other shall not be construed as a waiver of any succeeding breach of the same or other provisions, nor shall any delay or omission on the part of either party to exercise any right that it may have under this Licence operate as a waiver of any breach or default by the other party.

7. NOTICES

Any notices or instruction to be given hereunder shall be delivered or sent by first-class post or telecopier to the other party, and shall be deemed to have been served (if delivered) at the time of delivery or (if sent by post) upon the expiration of seven days after posting or (if sent by telecopier) upon the expiration of twelve hours after transmission.

8. ASSIGNMENT AND SUB-LICENSING

The Licensee may at his discretion assign the System and in doing so shall assign this Licence its rights and obligations to the purchaser who shall without reservation agree to be bound by this Licence. The original Licensee and any subsequent Licensees shall be bound by the obligations of this Licence in perpetuity.

9. LIMITATION OF LIABILITY

The Company's maximum liability under any claim including any claim in respect of infringement of the intellectual property rights of any third party shall be, at the option of the Company either:

(a) return of a sum calculated as the price received for the System by the Company from its immediate customer depreciated on a straight line basis over a one year write-off period; or

(b) repair or replacement of those components of the System that do not meet the warranties contained within this Document.

The foregoing states the entire liability of the Company to the Licensee.

10. CONSEQUENTIAL LOSS

Even if the Company has been advised of the possibility of such damages, and notwithstanding anything else contained herein the Company shall under no event be liable to the Licensee or to any other persons for loss of profits or contracts or damage (whether direct or consequential) arising in connection with the System or any modification, variation or enhancement thereof and including any documentation or data provided by the Company or for any other indirect or consequential loss.

11. ENTIRE AGREEMENT

The Company shall not be liable to the Licensee for any loss arising in connection with any representations, agreements, statements or undertakings made prior to the date of supply of the System to the Licensee.

12. TERMINATION

This Licence may be terminated forthwith by the Company if the Licensee commits any material breach of any terms of this Licence. Forthwith upon such termination the Company shall have immediate right of access to the System for the purpose of removing it.

13. SEVERABILITY

Notwithstanding that the whole or any part of any provision of this Document may prove to be illegal or unenforceable the other provisions of this Document and the remainder of the provision in question shall remain in full force and effect.

14. HEADINGS

The headings to the Clauses are for ease of reference only and shall not affect the interpretation or construction of this Document.

15. LAW

This Document shall be governed by and construed in accordance with English law and all disputes between the parties which cannot be resolved by negotiation shall be determined by arbitration in England in accordance with the Arbitration Act 1950 and 1979.

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PART 1

Operations manual

INTRODUCTION

Thank you for purchasing this CEDAR Series X+ audio restoration module.

It is one of a family of the world's most advanced and most effective noise removal units, and it offers outstanding processing power and performance. The full range consists of the:

- **BRX+ debuzzer**
  Removes a wide range of buzzes and hums from any audio material;

- **AZX+ azimuth corrector**
  Corrects the time-alignment of stereo and double-mono audio signals.

Quality, speed and simplicity are paramount considerations in the Series X+ design, and all the units in the family are designed for professional use. Additional Series X+ features include the following:

- **Flexibility**
  Each model in the Series X+ range will handle a wide range of audio restoration requirements.

- **Portability**
  Each Series X+ is a compact, lightweight unit designed to fit a standard 1U rack space.

- **Audio interfaces**
  Series X+ incorporates a 24-bit digital audio interface conforming to both AES/EBU and SPDIF standards.

- **Power**
  Its universal power supply means that a Series X+ will work anywhere in the world.

- **Processor**
  A 40-bit floating point DSP processor delivers 60MFlops so that the Series X+ will handle the most complex processing requirements.

SAFETY INSTRUCTIONS

Read these instructions, follow them, and save them for future reference.

- **Water and moisture**
  The unit must not be exposed to rain or moisture. Furthermore, if the unit is brought directly from a cold environment into a warm one, moisture may condense inside it. This, in itself, will not cause damage, but may cause electrical shorting. This could damage the unit, and even cause danger to life. ALWAYS allow a Series X+ to naturally reach ambient temperatures before connecting the mains power.

- **Mounting and ventilation**
  The unit may be mounted in a 19" EIA rack, or placed on a flat, stable surface. Do not subject it to strong sunlight, excessive dust, mechanical vibration or periodic shocks. The Series X+ is not susceptible to heat build-up, but should be installed away from heat sources such as radiators, and audio devices such as amplifiers that produce large amounts of heat.

  If it is used as a free-standing unit, the supplied rubber feet should be fixed to the base of the unit.

- **Power sources**
  The Series X+ features a universal power supply that will work safely on any mains supply in the ranges 85V to 260V, 50Hz or 60Hz AC only. The unit should always be grounded (or 'earthed'), and power connectors should be routed so that they will not be walked on or pinched.

  The Series X+ is connected to the mains power as long as it is plugged in to a mains supply, so if it is not to be used for an extended period, unplug it from the wall. Pull the connector out by the plug, never by the cord itself.

- **Connections**
  Turn off the power to all equipment before making any connections.

- **Cleaning**
  Clean the unit only with a dry cloth. Never use abrasive pads or liquid cleaners such as alcohol or benzene.

- **Damage requiring service**
  The Series X+ contains no user-serviceable parts and should on no account be opened or dismantled by unauthorised personnel.

  It should be returned to qualified service agents when it has been exposed to liquids, when it fails to function correctly, when it has been dropped, or when the case is damaged.

SETUP

Unpacking

Unpack the Series X+ carefully. Save the carton and all packing materials since you may need them to transport the unit in the future. In addition to the Series X+ and the packaging, the carton should contain the following:

- mains connection lead
- this manual
- rack mounting hardware
- four self-adhesive feet.

Installation site

To maintain reliability and prolong operating life, observe the following environmental considerations:

- the nominal temperature should be maintained between 5° and 35° Celsius
- relative humidity should be in the range 30% to 80% non-condensing
- strong magnetic fields should not exist nearby.

AUDIO CONNECTIONS

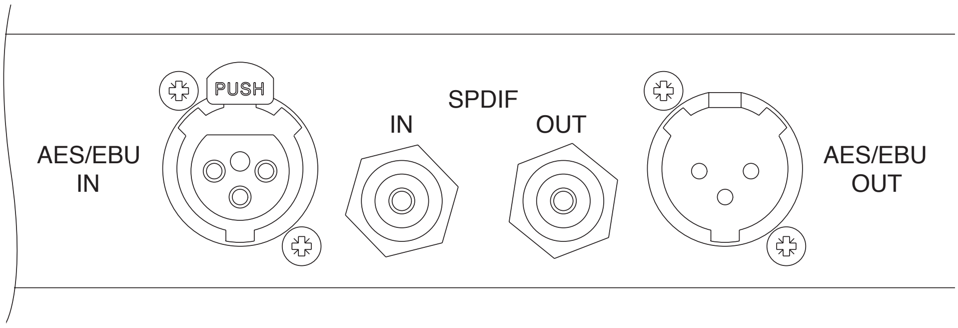
The Series X+ offers two audio connection standards, and passes its signal to both outputs irrespective of the input used. The standards are:

- **Digital SPDIF format audio data**
  SPDIF is used by domestic and semi-professional digital audio devices.
  You should connect the SPDIF output from your source to the SPDIF input of the Series X+ using a single cable terminated with an RCA (or 'phono') plug.

  The SPDIF output of the Series X+ should be connected to the SPDIF input of a recording device or external DAC.

- **Digital AES/EBU format audio data**
  The AES/EBU format is used by professional digital audio devices.
  You should connect the AES/EBU output from your source to the AES/EBU input of the Series X+ using a single cable terminated with an XLR plug.

  The AES/EBU output of the Series X+ should be connected to the AES/EBU input of a recording device or external DAC.



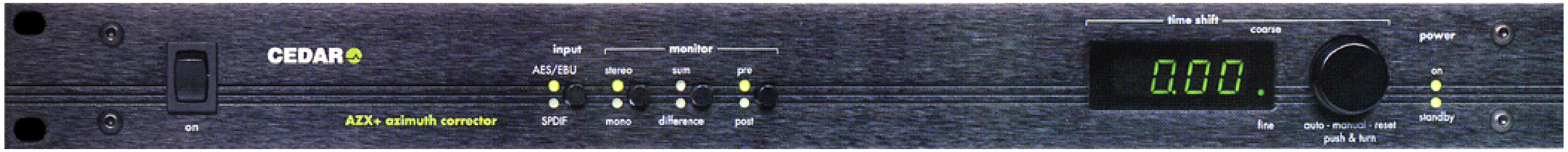
Channel status

No Series X+ processes are affected by channel status data. All units will echo any such data directly to the outputs.

CEDAR BRX+ debuzzer

1 2 3 4 10 11 12 13 14

CEDAR AZX+ azimuth corrector



1 2 5 6 7 8 9 14

FRONT-PANEL INDICATORS AND CONTROLS

1 Power switch

- When OFF, the Series X+ is in standby mode.
- When ON, the Series X+ is powered up and operational.

The unit is not powered off completely unless the power lead is removed.

2 Input selector and LEDs

- When OUT, the AES/EBU input is selected.
- When IN, the SPDIF input is selected.

If a Series X+ fails to lock to an incoming signal, the LED will flash.

3 BRX+ Output attenuator and LEDs

- When OUT, no attenuation is applied.
- When IN, the signal is attenuated by 3dB.

Use of the BRX+ processor on signals close to full-scale may result in slight clipping. This can be avoided using the attenuator.

4 BRX+ Process ON/OFF and LEDs

- When OUT, the input signal is echoed at the outputs.
- When IN, the processed signal is directed to the outputs.

Use this control for before/after monitoring of the signal. The signal delay and attenuation are not affected by this control.

5–13 Process controls

Refer to the section specific to each model: the section on the BRX+ starting on page 9, and on the AZX+ starting on page 13.

14 Power indicator

- Upper LED illuminated indicates that the unit is operational.
- Lower LED illuminated indicates that the unit is in standby mode.

Either LED illuminated indicates that power is applied to the unit.

The BRX+ debuzzer

OPERATING THE BRX+ DEBUZZER

Introduction

CEDAR's buzz and hum removal process uses information about the fundamental frequency and the overall volume of the buzz or hum to identify, up to a limit of 4kHz, all the harmonics that constitute the noise. It is left to the user to determine how much of that noise is then removed.

Fundamental frequency

Your first job will be to identify the fundamental frequency of the buzz or hum. This is accomplished most easily by setting the PROCESS button to ON and then rotating the BUZZ POWER and ATTENUATION knobs to their maximum values (fully clockwise). There may be artefacts such as a 'comb-filtering' effect at this point. You can safely ignore these because they are a consequence of the over-processing being applied by these maximum settings.

Note: If the buzz is very low in level compared with the rest of the signal, you should initially set the BUZZ POWER to '12 o'clock' rather than to maximum.

If the fundamental frequency is set wrongly then the process will not be able to identify the buzz.

The FREQUENCY knob has three modes of operation: preset value selection, fine continuous adjustment, and coarse continuous adjustment. You may select between them as follows:

- **Preset selection**
  The preset frequencies represent the most commonly encountered fundamental frequencies. You can select between these by pressing the FREQUENCY knob and keeping it depressed while you rotate it.

  In cases where the true fundamental frequency does not lie precisely on a preset frequency, you should use the FINE adjustment mode to adjust the value.

- **Fine adjustment**
  Pressing and releasing the FREQUENCY knob toggles between the COARSE and FINE (0.01Hz below 100Hz, 0.1Hz above 100Hz) adjustment modes. The COARSE and FINE indicators in the FREQUENCY display will tell you which mode you have selected.

Coarse adjustment

If you do not know the fundamental frequency, you will need to 'hunt' for it. Alternatively, you may know the frequency, but it does not lie close to one of the presets.

In both these cases you will wish to adjust the FREQUENCY control in coarse steps. Pressing and releasing the FREQUENCY knob toggles between COARSE and FINE (0.01Hz below 100Hz, 0.1Hz above 100Hz) adjustment modes. The COARSE and FINE indicators in the FREQUENCY display will tell you which mode you have selected.

If you are hunting for the fundamental frequency, best results are usually obtained by starting at the maximum frequency (130Hz) and scrolling slowly down through the range until you hear the buzz or hum become significantly attenuated.

Note: You must be careful to find the true fundamental frequency. If the buzz has a 50Hz fundamental frequency, a frequency setting of 100Hz might possibly be of some audible benefit: it would attenuate the even harmonics, but it would not attenuate the fundamental itself or any of the odd harmonics. Similarly, a setting of 25Hz might have some beneficial effect, but it might damage the true signal at 25Hz, 75Hz, 125Hz, and so on. In both cases the processing would be more effective if 50Hz was correctly selected.

Tracking indicators

The BRX+ searches for a buzz fundamental in a small window centred on the displayed fundamental frequency. The width of this window is approximately 1Hz at a setting of 50Hz, and increases with increasing frequency. If a consistent buzz is detected within this window, one or more of the tracking indicators will light.

The top tracking indicator LED is a red, downward-pointing arrow which lights if the displayed fundamental frequency is higher than the detected buzz frequency. The bottom indicator LED is an upward-pointing arrow which lights if the displayed fundamental frequency is lower than the detected frequency.
If either of these LEDs is lit, you should use the FINE adjustment mode to tune the fundamental frequency until the central green LED lights to indicate that the algorithm is optimally set.

The tracking system takes approximately two seconds to settle, and will thereafter automatically track slow fluctuations of the buzz, provided that it remains within the window. If no consistent buzz is detected in the window, any or all of the LEDs will flicker as the algorithm continues to 'hunt'. Loud musical components that fall within the window can upset the operation of the tracking system; this effect is minimised by correct setting of the BUZZ POWER control.

Buzz power

Once you have correctly identified the fundamental frequency, the buzz or hum should be inaudible or, at the very least, greatly reduced. However, the BUZZ POWER and ATTENUATION controls have not yet been adjusted so it is likely that the material is being over-processed.

Your next step, therefore, is to reduce the BUZZ POWER control to a level just above that at which the buzz becomes audible again.

Attenuation

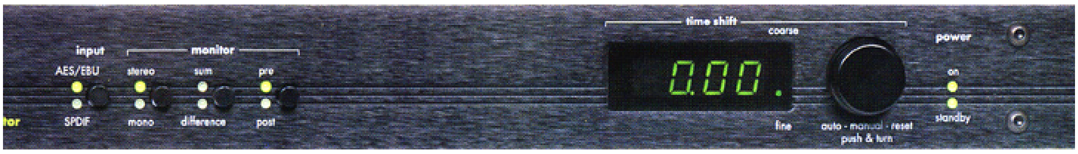
If both the FREQUENCY and BUZZ POWER controls are set correctly, you will have given the BRX+ all the information it needs to correctly identify the buzz or hum, both in terms of its frequency spectrum and its amplitude. It now remains only to determine how much of the noise is removed from the signal.

Your final step, therefore, is to reduce the ATTENUATION control (turn it anti-clockwise) to a level just above that at which the buzz becomes audible. In doing so, you will minimise the possibility of any signal being removed along with the buzz or hum.

Note: There are occasions where a signal suffers from two or more buzzes or hums and in these cases it will be impossible to remove the problem in a single pass of the audio. If you encounter a signal damaged in this way, you should perform multiple passes using the BRX+, or a single real-time pass on a CEDAR for Windows system.

The AZX+ azimuth corrector

OPERATING THE AZX+ AZIMUTH CORRECTOR



Introduction

CEDAR's Phase/Time correction process uses information common to the left and right channels of a stereo signal to determine the offset between the channels. It then recreates the signal with sub-sample accuracy so that the channels are correctly aligned.

The process has two primary stages:

- You or the AZX+ determines the timing error
- The AZX+ corrects the timing error.

To aid rapid setup and accurate processing you can monitor the output:

- before or after processing
- in stereo
- with the left and right channels summed (mono sum)
- with the right channel subtracted from the left (mono difference).

The display indicates the time shift being applied to the signal. A positive number indicates that the right channel is being delayed with respect to the left; a negative number indicates that the right channel is being advanced relative to the left.

The principal control on the AZX+ is the ADJUST knob, located to the right of the TIME SHIFT display.

Modes of operation

The AZX+ offers two primary modes of operation – AUTO and MANUAL – plus a quick RESET facility:

- **AUTO mode**
  Press and rotate the ADJUST knob until the screen flashes the word 'Auto'. When you release the ADJUST knob, the central tracking light (in the far left of the display) will flash and the unit will be in AUTO mode.

- **MANUAL mode**
  Press and rotate the ADJUST knob until the screen flashes the word 'USEr'. When you release the ADJUST knob the unit will be in MANUAL mode.

- **RESET**
  Press and rotate the ADJUST knob until the screen flashes the word 'ZEro'. When you release the ADJUST knob, the value in the TIME SHIFTER will be reset to zero. The tracking mechanism will also be reset.

AUTO mode

In AUTO mode, the AZX+ measures the offset between the channels and automatically applies a compensating correction to the signal. It is not possible to manually alter the amount of correction in AUTO mode. The central tracking light flashes to indicate that AUTO mode is engaged.

MANUAL mode

If you select the MANUAL mode you will be able to set manually the amount of time correction applied to the signal, with a resolution of up to 1% of a sample. You do this by rotating the ADJUST knob until the desired time shift is shown in the display.

- **Coarse and fine adjustment**
  If, once in MANUAL mode, you press the ADJUST knob and then release it without rotating it, you will toggle between the COARSE and FINE adjustment modes. The COARSE and FINE indicators in the TIME SHIFT display will tell you which mode you have selected.

  In the COARSE mode you will shift the timing offset in steps of one sample.

  In the FINE mode you will shift the timing offset in steps of one hundredth of a sample (up to 9.99 samples) or one tenth of a sample (from 10.0 to 99.9 samples).

Note: An integer time shift is equivalent to using an editor to record your signal, removing or inserting samples at the start of either the left or right channel, and then playing back the file.

- **Tracking lights (MANUAL mode)**
  When adjusting the time shift in MANUAL mode you may wish to monitor the tracking lights on the far left of the TIME SHIFT display. These are telling you whether the manually selected time shift is greater than, less than or equal to that determined by the autotracking algorithm.

- If the red, upward-pointing arrow is lit, you can turn the ADJUST knob clockwise to move closer to the autotracker value.
- If the downward-pointing arrow is lit, you can turn the ADJUST knob anticlockwise to move closer to the autotracker value.

Note: If the central green LED is lit alone then the displayed TIME SHIFT is within a quarter of a sample of the detected value. If an arrow and the central LED are lit together then the displayed value is within one sample of the detected value. If an arrow is lit alone then the displayed and detected values differ by more than one sample.

Reset

To reset the AZX+, press the ADJUST knob, rotate it until 'ZEro' is displayed, and then release it.

Monitor modes

Once you have determined the correct timing correction, the AZX+ will process the signal according to the following monitor mode controls:

Monitor mode: PRE/POST

- **POST**
  The time shift shown in the display is applied to the signal.

- **PRE**
  The value shown in the display is ignored, and the signal is unaffected by the TIME SHIFTER.

  This control allows you to make quick, glitch-free A/B comparisons of the signal before and after azimuth correction.

Monitor mode: STEREO/MONO; SUM/DIFFERENCE

- **STEREO**
  The left and right signals from the TIME SHIFTER are passed to the left and right outputs.

- **MONO**
  The left and right signals from the TIME SHIFTER are either summed, or the right channel is subtracted from the left channel, according to the position of the sum/difference control. The resulting monophonic signal is passed to the left and right outputs.

Note: If the original audio is a true mono signal with the left and right channels in perfect alignment, the result of selecting MONO/DIFFERENCE will be perfect cancellation (silence).

Note: Since SUM and DIFFERENCE are not applicable to STEREO mode, the control is disabled when STEREO is selected.

Tutorial

If AUTO mode is selected and the unit is in PRE, the AZX+ will act purely as a phase meter, and it will then be apparent when you have timing errors in any material you pass through it.

If you detect timing errors and wish to correct them, the suggested procedure is as follows:

1 Ensure that the AZX+ is in POST.

2 Select AUTO and STEREO modes. If the result is satisfactory you need proceed no further.

Sometimes the material will contain information that hinders the operation of the autotracker, and this can cause the TIME SHIFT to swing or 'hunt' for the correct value. If this happens, you should use MANUAL mode.

3 Put the AZX+ into MANUAL, MONO, and DIFFERENCE modes.

4 Sweep the time relationship using the ADJUST control. Do so until you find the TIME SHIFT at which the perceived signal is minimised.

Hint: You can often recognise this by noticing that vocalists and solo instruments disappear from the mix, leaving just the reverberation behind.

5 Return to STEREO mode, and use PRE/POST to compare the uncorrected and corrected signals.

6 If you are satisfied with the results, restart the audio material and allow the AZX+ to process it.

Note: You may adjust the TIME SHIFT manually while the material is playing without introducing glitches. This may be of use when you wish to use MANUAL mode but know that there are a limited number of changes in the TIME SHIFT during the course of the material.

Note: You may use MONO SUM mode to assess the mono-compatibility of the processed and unprocessed signals. In this mode, the level is reduced by 6dB to avoid the possibility of signal clipping.

SPECIFICATIONS

General

Power supply: 85–260VAC; 50–60Hz

Power consumption: 15W (standby 1W)

Weight: 2.5kg (net); 3.5kg (gross)

Overall dimensions: 45 x 483 x 240mm

Audio

I/O type: Digital PCM

I/O resolution: 24 bits

Sample rate: 30–50kHz

Data format: SPDIF or AES/EBU

Processor: 60 Mflops

Group delay (milliseconds)

	BRX+	AZX+
44.1kHz	40.8	49.5
48kHz	37.5	45.5

EMC REGULATIONS

In order to fully comply with EMC regulations, the Series X+ should be connected using metal-shelled connectors and good quality shielded cable suitable for digital audio.

Declaration of conformity

Date of issue	1 February 1999
Equipment	CEDAR 'Series X+'
Manufacturer	CEDAR Audio Ltd
Address	9 Clifton Court, Cambridge, CB1 7BN, UK

This is to certify that the aforementioned equipment, when used in accordance with the instructions in this manual, fully conforms to the protection requirements of the following EC Council Directives: on the approximation of the laws of the member states relating to:

- **89/336/EEC Electromagnetic Compatibility**
  Applicable standards: EN 50081-1:92
  EN 50082-1:92

- **73/23/EEC Low Voltage Equipment**
  Applicable standard: BSEN 60-065:1994

PART 2

An introduction to audio restoration

BUZZ REMOVAL

Buzz is a word that we use to refer to many audio degradations. It sometimes describes the noises produced by, for example, problems with lighting rigs and faders. These noises often take the form of closely spaced, regular ‘ticks’ in the signal. On other occasions, ‘buzz’ is used to describe the sounds produced by electrical faults such as earthing problems. These can lead to a variety of unwanted noises such as simple mains hum or harmonically complex tones. (Figure 1).

You can view and modify audio signals in two principal ways – in the time domain (in which you consider the waveform of the signal), and in the frequency domain (in which you consider the spectral content of the signal). It is important to remember that these domains are just different ways of considering the same information.

Nevertheless, it is most effective to handle the audio and modify it in the domain that is most appropriate to the problem. Therefore, for example, a buzz that is composed of closely spaced, regular ‘ticks’ is best restored using one of CEDAR’s various decrackle (time domain) processes, and we will not consider it further here. On the other hand, a buzz that has an indistinct time domain structure may be more easily characterised by its harmonic content and then analysed and treated in the frequency domain.

Historically, audio engineers have used a number of conventional filtering techniques to eliminate buzzes and hums. The simplest of these is the basic high-pass filter, which attenuates the signal at all frequencies below a specified cut-off point. (Figure 2). This can be used to eliminate harmonically simple hums which, because of the mains frequencies of most countries, have fundamentals of 50Hz or 60Hz. Unfortunately, such a filter will also eliminate some of the desired sound as well as the unwanted tone, resulting in a gutless signal that lacks bass.

Some complex buzzes and hums exhibit harmonics reaching up to many kHz. But these too have significant energy at lower frequencies, so a low-pass filter is completely inappropriate for removing buzz. A more sophisticated approach involves the use of comb-filters – so-called because the response of the filter resembles the teeth of a comb. (Figure 3).

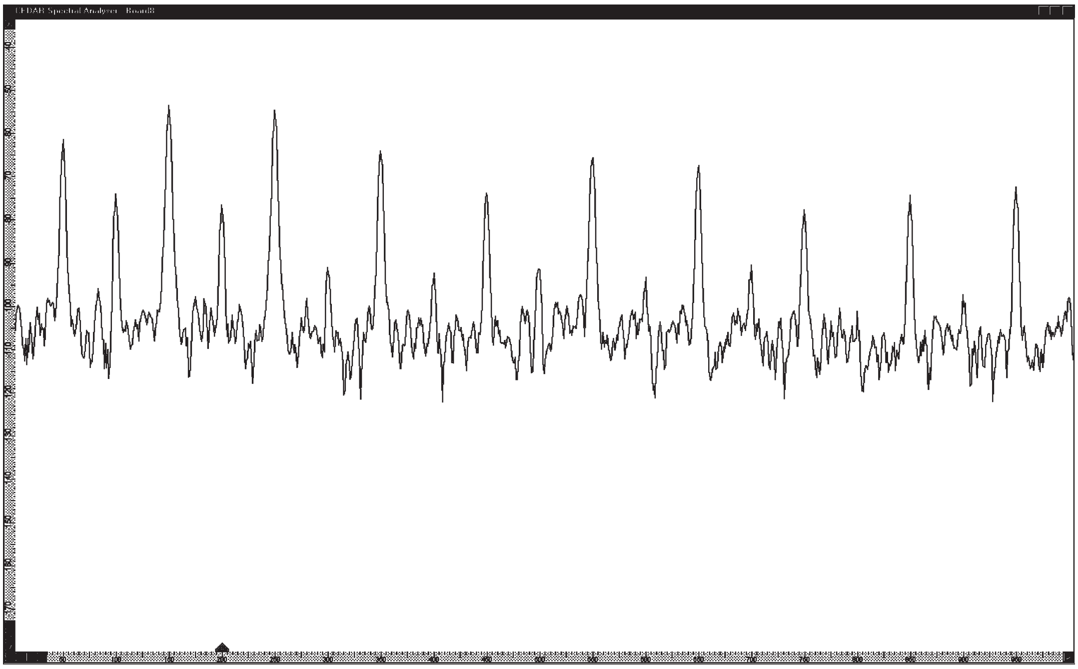


Figure 1

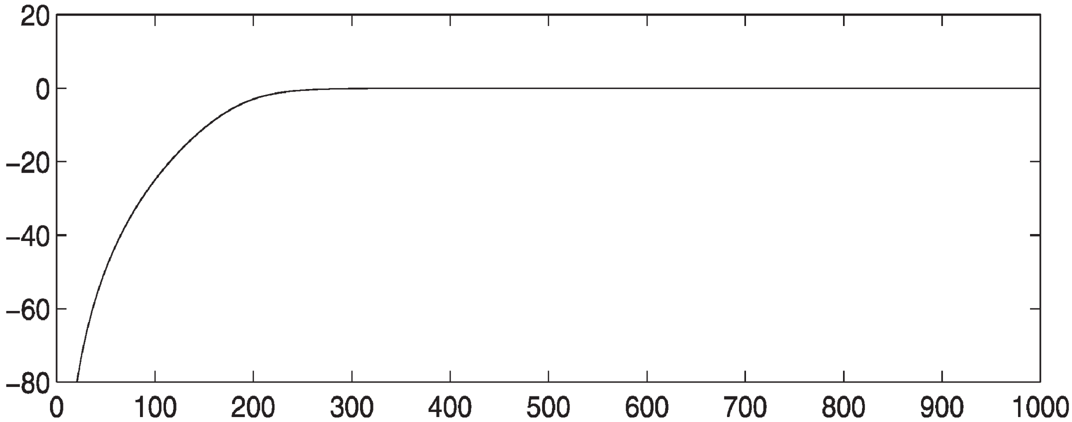


Figure 2

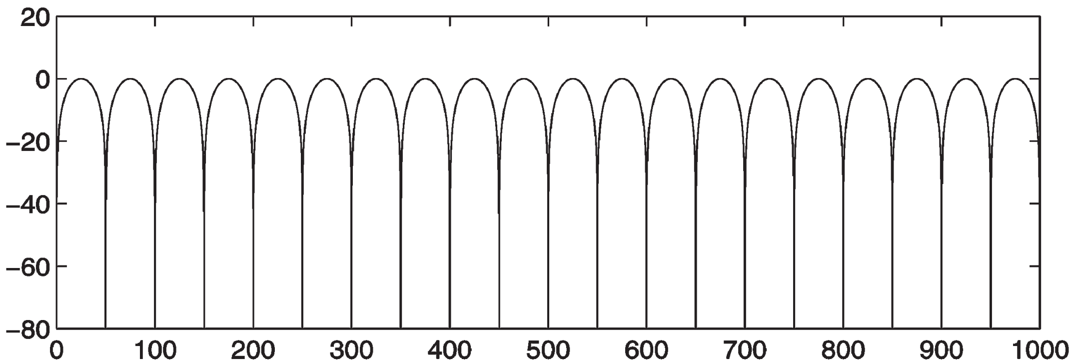


Figure 3

These filters allow you to remove signal components at a number of regularly-spaced frequencies. Furthermore, the tighter the ‘teeth’ of the comb are, the more precise the removal can be. This looks, at first sight, like an ideal solution, but it isn’t. For one thing, the filter exhibits the same behaviour as all other static filters: if, at a given frequency, you remove an offending noise, you also remove any genuine signal that may exist at the same point in the spectrum. (Figure 4).

Secondly, comb-filters introduce a characteristic ‘boxiness’ that makes the genuine signal sound distant and indistinct. This problem becomes worse the more precise the ‘teeth’ become so, in limiting the unwanted side effects caused by the first problem, you exacerbate the second. Thirdly, the offending frequencies may not be entirely constant, so the filter’s ‘teeth’ may not always be appropriately placed, and you will hear the buzz come and go during the course of the material you are playing.

These problems make comb-filters – which are, nonetheless, implemented on a number of audio processing platforms – unacceptable for the highest quality removal of buzzes and hum. A far better solution is an algorithm that identifies the offending frequencies, tracks any fluctuations that they experience, and then removes the noise without destroying the underlying signal. (Figure 5).

CEDAR’s Debuzz process precisely fulfils this specification. It will track buzz frequencies within a range of 2% at 50Hz, and will retain genuine signal components while removing any buzz harmonics that exist at the same frequencies.

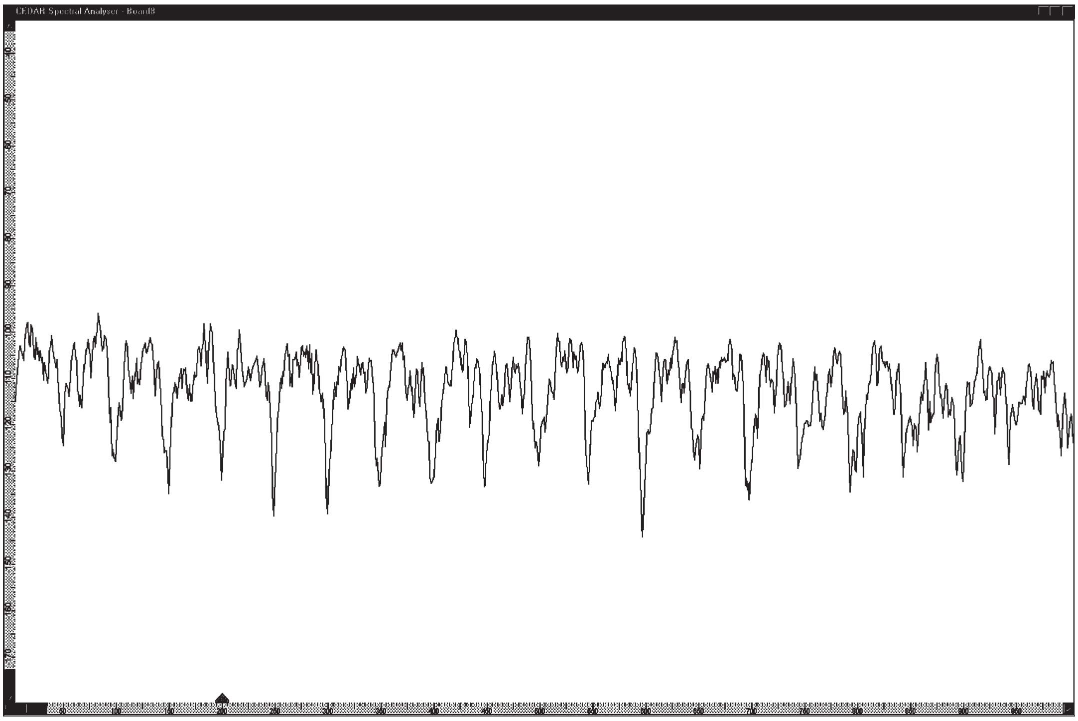


Figure 4

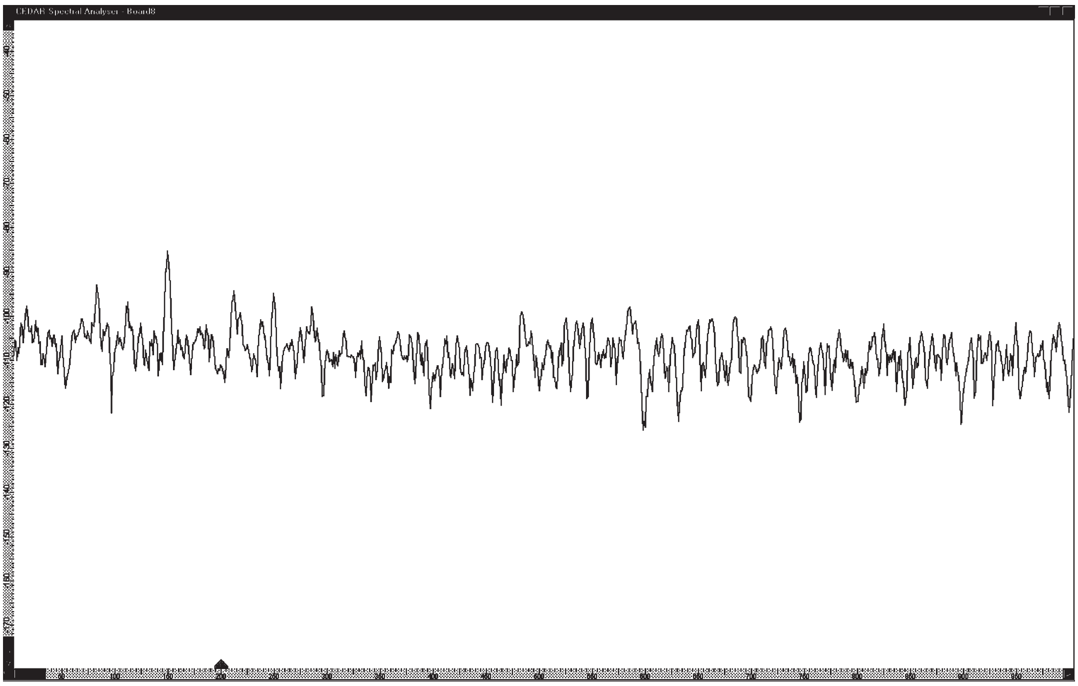


Figure 5

AZIMUTH CORRECTION

CEDAR's first version of the Azimuth Corrector was called the Phase/Time Corrector, and ran on the CEDAR-2 Production System, an early PC-based suite of processes that ran under DOS. This was developed primarily for the broadcast and video industries, for whom mono compatibility is of great concern. The name was, however, misleading because the algorithm did not correct 'phase' errors within the signal. It corrected timing errors.

The next incarnation of the process was implemented in the AZ-1 Azimuth Corrector, a CEDAR Series 2 rackmount unit. The reason for the change of name – despite the functionality remaining broadly unchanged – was purely a marketing decision. Audio engineers knew what azimuth errors were, and the unwanted side effects that they created.

Unfortunately, the use of the term 'azimuth' implies that these errors only occur when analogue tape recorders are involved in the signal chain. This is not the case. Indeed, timing errors can occur whenever two signal paths are inconsistent with each other. They can even occur – although to a lesser extent – if the pick-up cartridge in a turntable is misaligned. Nevertheless, this is the name that makes most sense to the world at large, and that is why the AZX+ is called an Azimuth Corrector rather than a stereo timing corrector.

To understand what these timing errors do to a signal, let us first consider some simple cases involving just sinewave oscillators. A sinewave oscillator can be described in its entirety by just three parameters: its frequency, amplitude, and phase. The frequency and amplitude describe, to a good approximation, the perceived pitch and volume of the signal. The phase, however, is of no audible significance until we combine more than one such sinewave.

For the moment, let us consider two sinewaves of the same frequency and amplitude, but different phases. Figure 1 shows a simple sinewave climbing away from 'zero' at T=0. The figure also shows another, identical waveform with identical phase (i.e. starting at the same time). As you would imagine, the two waveforms add together to produce the same sound, but louder.

But now consider figure 2. This also shows a simple sinewave climbing away from 'zero' at T=0, with another, identical waveform offset by half its cycle. If we add these two waves together, they cancel each other out, and we hear nothing. Although, in isolation, both oscillations sound identical, combining them results in silence.

This is a very simple result, and demonstrates perfect addition and perfect cancellation. These can also be represented by the Lissajous diagrams shown in figure 3 (in phase), and figure 4 (out of phase). If you combine the waveforms with other delays (i.e. at different phases) you get other results that lie between the maximum volume and silence, and the Lissajous figure becomes an elipse. What's more, if, instead of combining these waves into a single signal, you output them through different speakers, you hear a different result. When the signals are in phase you hear the original tone, perfectly reproduced in mono. But as the phase shifts, you hear the tone shift across the sound field. (Figure 5.)

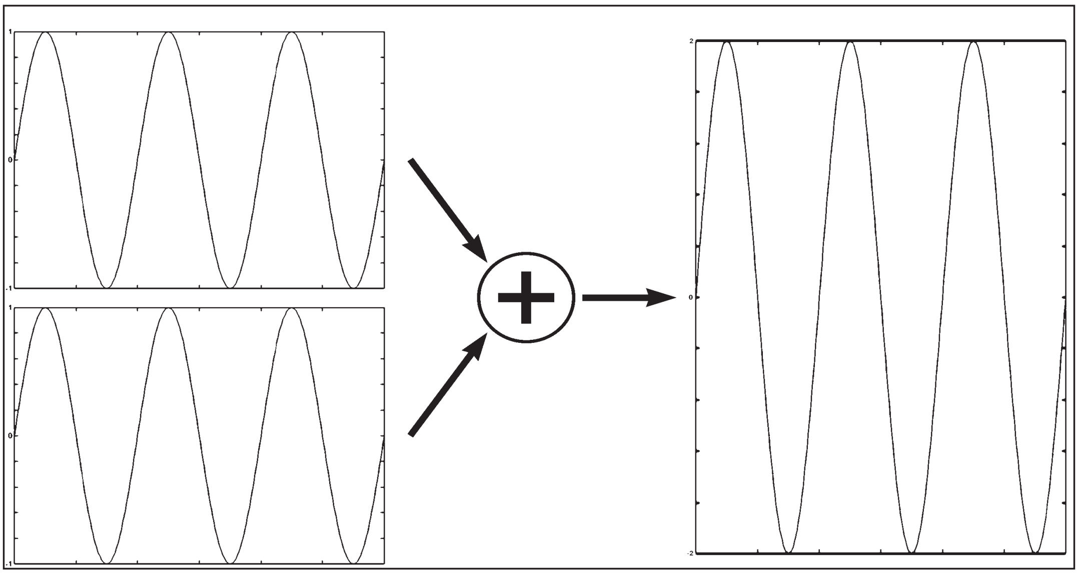


Figure 1

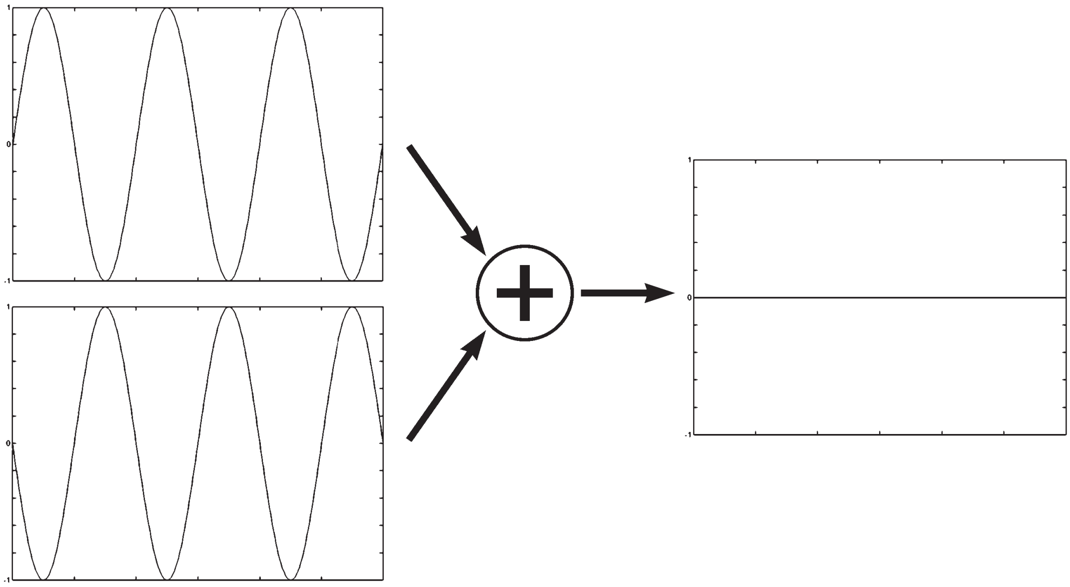


Figure 2

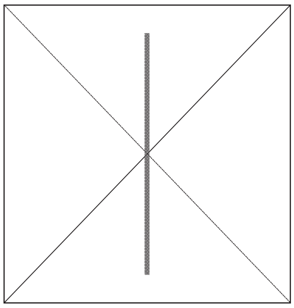


Figure 3

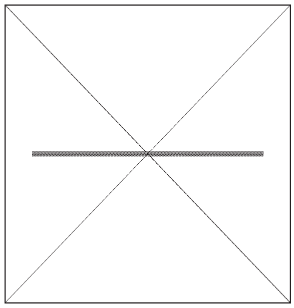


Figure 4

Of course, the relationship between the phase shift and the timing error is defined by the frequency of the signal. If the sinewave frequency increases or decreases, a given timing difference will lead to different relative phases between the signals. This concept then leads to the following situation.

Consider two complex signals, such as those that represent music or speech. Fourier analysis tells us that these can be represented by an infinite number of sinewaves that represent all the frequencies present in the signal. So, for any given timing difference between the left and right channels, each sinewave in the signal will be phase-shifted by a different amount. Some will be added together and thus become louder, most will be smeared across the stereo image, and some will (if reproduced in mono) cancel out entirely. The result, when viewed on a spectral analyser, looks exactly like a broad comb-filter, with the distance between the 'teeth' of the comb defined by the size of the timing error. (Figure 6).

The distortion of these phase relationships and the consequent filtering effect cause all the audible problems of timing errors. They include loss of high frequencies, muddy bass, poor mono compatibility, and a general smearing of the stereo image. Worse still, if the timing error is not precisely constant, you will hear a 'flanging' effect caused by the 'teeth' of the comb filter sweeping backwards and forwards through the spectrum.

Audio engineers have traditionally employed a range of processors to hide these deficiencies: equalisers, stereo enhancers, dynamics processors and reverb units. However, none of these attacks the heart of the problem – these small but significant non-synchronisations of the left and right channels.

CEDAR corrects these problems by eliminating the timing errors that cause them. It does this by identifying any monophonic component in the stereo sound and then measuring the timing difference between the left and right channels of the signal.
If any such error is detected, the system recreates the signals such that they are accurately aligned. The timing measurement is performed nearly fifty times a second, and the signal is corrected dynamically according to each of these measurements. Zipper noise and digital glitching are avoided by regenerating the signal with an accuracy of 1% of a sample. Furthermore, a sophisticated filter minimises the chance that the detector will be fooled by unusual signals that contain little or no monophonic components, so that CEDAR should never generate erroneous corrections and introduce phase/time errors of its own.

There is one other use for CEDAR's phase/time correction products, and this is related to broadband noise reduction. If you transcribe a monophonic signal held on analogue tape or disc using a stereo tape-head or cartridge, you will create two monophonic signals. Each of these will contain the desired signal, and some noise. If you then time-align these channels and sum them to mono using CEDAR, you will generate a monophonic signal in which the desired signal is 6dB louder than either of the individual channels, but in which the broadband noise content may be as little as 3dB louder. (This is a consequence of the random nature of broadband noise.) As a result, you have given yourself a head start of (up to) a 3dB increase in signal-to-noise ratio before you attempt noise reduction on the signal.

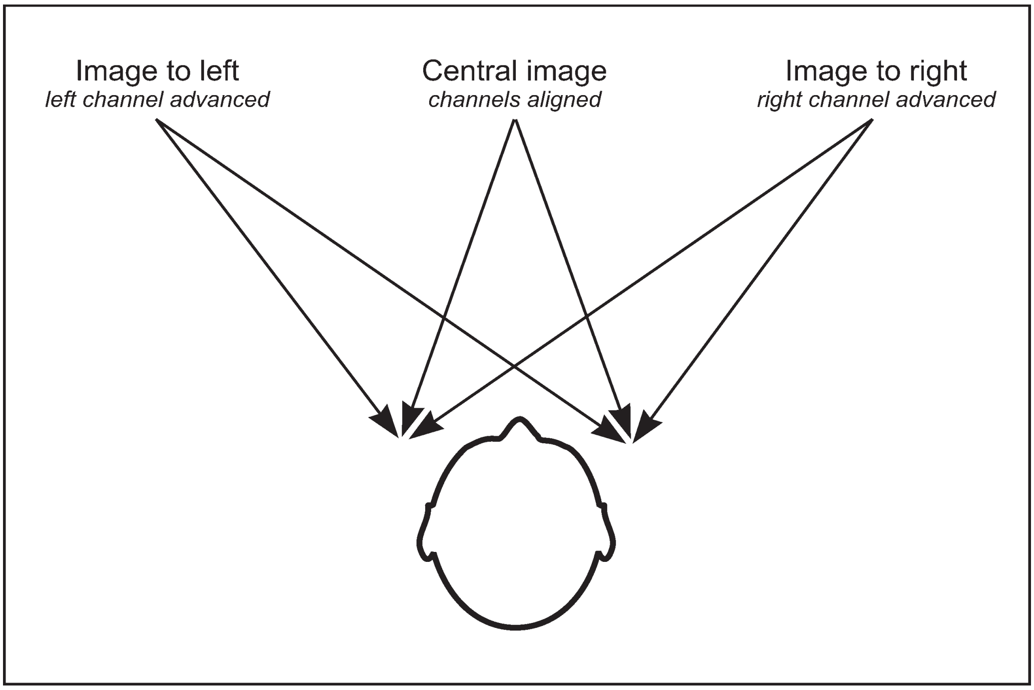


Figure 5

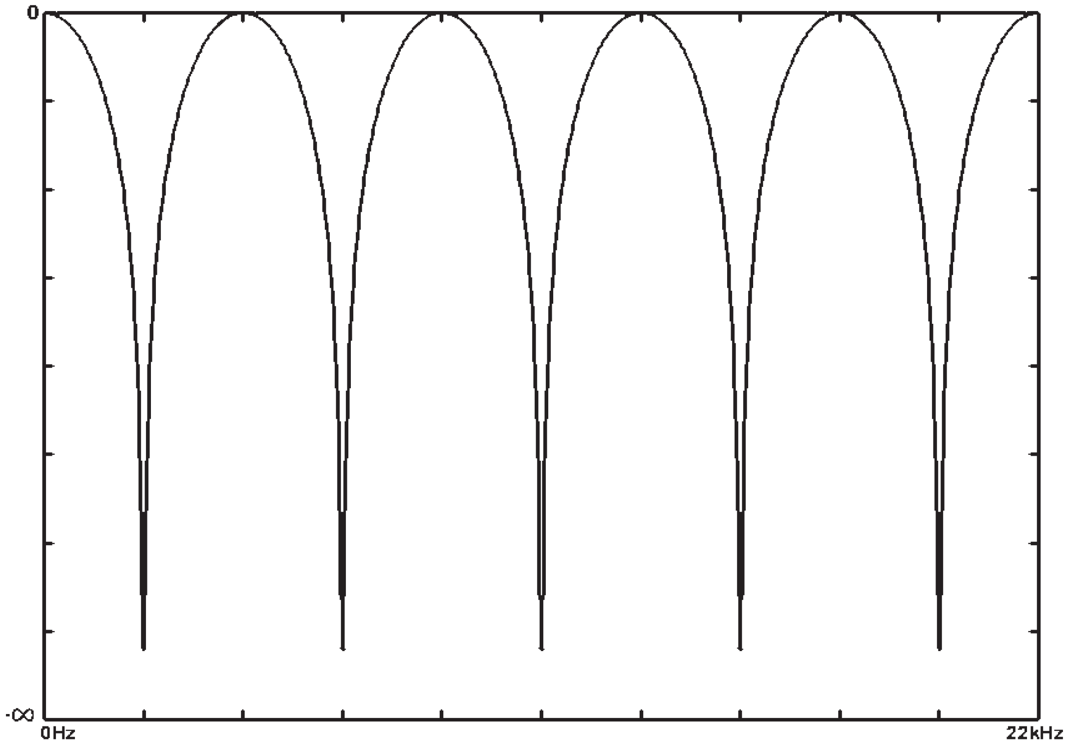


Figure 6

Tape-head azimuth errors

When you retrieve the sound from a stereo tape that has been either recorded or replayed off-azimuth, there are two sources of audible degradation.

- The first effect is caused by the distance – in the direction of tape travel – between the centres of the tape heads. This is the primary timing difference, and it is this that CEDAR's processes detect and correct.
- The second effect is the smearing caused by the length of tape covered – again in the direction of tape travel – by the offset head. In general terms, the head will then 'average' the signal lying underneath it at any point rather than measuring the instantaneous value. The degradation caused by this will be smaller than that caused by the primary error, and the effect will therefore be less significant. However, it will lead to a loss of high-frequency content, and to a small reduction in dynamic content.

CEDAR Series X+

Serial number:	
Inspected:	
QC Engineer:	

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