

User's Guide



ADI-96 PRO

CDS[®]

Constant Double Speed

Analog Processing

Digital Processing

Hi-Precision

24 Bit / 96 kHz Stereo

DSP based Analog / Digital-Converter

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1. Introduction

Congratulations on your purchase of a ADI-96 PRO. This hi-quality analog to digital converter uses the unique *Constant Double Speed* technology to precisely convert analog audio data into different digital audio formats. Newest circuit technology combined with modern integrated circuits resulted in a unique and outstanding device, meeting highest quality standards. The ADI-96 PRO will excite you even after many years of operation.

2. Supplied Contents

Please ensure that all the following parts are included in ADI-96 PRO's packaging box:

- ADI-96 PRO
- Manual
- Power cord
- Serial cable 9-pin D-type
- RME Driver CD including Windows and BeOS software *Remote Control*

3. Brief Description and Characteristics

The ADI-96 PRO is a 2-channel analog to digital converter in a 19" rackmount enclosure of 1 U height. Newest 24 bit converters using 64 times oversampling result in 115 dBA dynamic ratio. This value is not only printed in the brochure, thanks to our Low Jitter Design it is available with every sold unit.

The servo balanced line inputs are fitted with both XLR and 1/4" TRS jacks separately. The signal path from the jacks to the ADC is totally balanced. A super low noise amplifier stage provides 20 dB adjustable gain, set by two hi-precision potentiometers on the front panel. This brings a perfect adaptation to the most used levels -10 dBV and +4 dBu for each channel separately.

The discrete low noise and low distortion microphone preamp works totally independent from the line circuits. A high quality relays connects the output of the line and mic electronics directly to the AD-converter. The Mic XLR jacks also provide +48 V of phantom power for condenser microphones.

The clearly structured front panel design ensures an easy start when working with the device for the first time. The Peak Level Meter is visualized by a hi-precision LED chain with reliable Over detection. The AD-converter is able to sample at 32, 44.1, 48, 64, 88.2 and 96 kHz. The digital output is available at SPDIF (coaxial and optical) and AES/EBU (XLR) connectors.

Thanks to the unique *Constant Double Speed* technology the crystal clear sound of 96 kHz sample rate is preserved even when using common sample rates below 64 kHz, as the ADI-96 PRO samples and calculates in double speed. RME's exclusive *Analog Processing* brings typical analog behaviour and sound to the digital domain. When using Look Ahead our *Digital Processing* technology minimizes changes in sound even with heavy processing.

The DSP adds phase shift, low cut and mid/side processing in pure digital 48 bit quality. Processing all data in the DSP avoids all unwanted effects typical for analog circuitry, like additional noise, distortions, changes in frequency response and deviations between both channels. Calculating Phase, Low Cut, Expander, De-esser, M/S-Processing and Compressor by a 48 bit/96 kHz DSP results in a previously unreached sound quality.

4. Technical Specifications

- Power supply: Internal, 110 / 230 V AC, 30 Watts
- Dimensions 483 x 44 x 205 mm
- Weight: 2 kg

4.1 Analog Specs

- Resolution AD: 24 bit
- Dynamic ratio: > 115 dBA
- THD+N: > 100 dB, < 0.001 %
- Maximum input level AD: +22 dBu
- Minimum Line input level for 0 dBFS: +2 dBu
- Minimum Mic input level for 0 dBFS: -50 dBu
- Frequency response AD, -0.1 dB: 10 Hz - 21 kHz (sf 44.1 kHz)
- Frequency response AD, -0.5 dB: 5 Hz – 44.8 kHz (sf 96 kHz)
- Input Line: XLR, 1/4" TRS, servo balanced
- Input impedance Line: 10 kOhm
- Input Mic: XLR, servo balanced, +48 V phantom power switchable
- Input impedance Mic: 2 kOhm

4.2 Digital Specs

- Super Low Jitter Design: < 3 ns in word clock PLL operation
- Output voltage: coaxial 0.5 Vpp, XLR 5 Vpp
- Output format: Consumer with/without copyright, Professional
- Supported sample rates: 32 kHz, 44.1 kHz, 48 kHz, 64 kHz, 88.2 kHz, 96 kHz
- Supported sample rates by word clock: 25 kHz - 105 kHz
- Internal resolution: 48 bit
- Resolution on digital output: 8, 16, 20, 24 bit
- Dither: 0 / 0.5 / 1 LSB triangular switchable
- Dither method: DC-offset free adding and rounding
- Output optical (TOSLINK), coaxial, XLR
- Format output SPDIF and AES/EBU (Consumer/Professional)
- Word clock input or output switchable
- Serial RS-232 interface

5. First Usage - Quick Start

Connect your analog signal source to the ADI-96 PRO's XLR or TRS input jacks. First adjust the knobs fully counterclockwise and try to achieve an optimum input level by adjusting the source itself. Fine adjusting the input level can easily be done by watching the Peak Level Meter while turning the knobs. The best input level is present when all LEDs except the Over LEDs are lit at the loudest parts of the signal. When there is no level indication at all, check the actual input state (Mic/Line) in menu INP for left and right channel.

The analog line inputs of the ADI-96 PRO can be used with +4 dBu and -10 dBV signals. The ADI provides (stereo) TRS and XLR jacks. Both are internally connected, so not operational at the same time. The electronic input stage is built in a servo balanced design which handles monaural and stereo jacks correctly. When used unbalanced it automatically corrects the gain by 6 dB.



When using unbalanced cables with XLR jacks we recommend to short pin 3 of the cable's jack to pin 1 (ground). Otherwise noise may occur, caused by the unconnected negative input of the ADI-96 PRO's input jack.

The Mic XLR jacks also provide +48 V of phantom power for condenser microphones (menu INP, Mic. Phantom Powered, separately for left and right channel).

When switched on for the first time the DSP software *Dual Mono A/D* is started. To change to another DSP software see chapter 7.4 'Boot other function'.

It is more than likely that when using the device for the first time some additional settings in the digital section have to be changed. The device is shipped with 16 presets, of which preset 1 and 2 are highly compatible to common equipment. Please choose correct sample frequency and format (Professional, Consumer) in the menu OUT, according to the capabilities of the device connected to the ADI-96 PRO's output.



Coaxial, optical and XLR jack of the digital output always carry the same signal. A connection to professional devices with AES/EBU inputs requires the channel status Professional. With this setting consumer DATs connected in parallel will not start to record, as they require the format 'Consumer'.

Transferring digital data into a PCI bus equipped computer is best done using RME's digital interface cards of the DIGI96[®] and Hammerfall[®] series. These hi-quality cards come with drivers for all popular operating systems. They have a world wide reputation as ultimate solution for master and multitrack tasks.

6. Operation and DSP Functions

The ADI-96 PRO's user interface is easy to understand thanks to its clearly structured front panel design. The big and bright 2 x 20 character LC-display plus a simple and straight cursor controlled menu structure will probably make you never look into this manual. Because of the numerous functions found in the ADI and its software, we strongly encourage you to study this manual.

Let's start with a short introduction into the ADI-96 PRO's user interface philosophy.

After power-on and automatic calibration the Info window will come up and show the currently loaded DSP software. The device remembers all settings even without saving them. To be sure a certain setup is being used unaltered, you should first reload it using *Recall* (see 6.9 Setup).

Press *Enter* to bring up the Overview window, which displays all configurations and the available DSP functions. Each function is represented by three letters, and is the highest menu of several submenus.

On this level each function can be activated and de-activated (Bypass) using the +/- keys. That's why they are also labeled *ON/OFF*. Below this row of functions two lines out of thin lines, fat lines and bars can be seen. These two lines represent one channel each. The state of each function is shown by a thin line (off) and a fat line (on). The screenshot below shows all functions activated except LOC (Low Cut). A global bypass, useful for comparing the sound with and without DSP functions can be set by pressing the additional key *Bypass DSP*.



To avoid pan shifts between left and right channel the control paths (sidechains) of some functions can be linked. As shown in the above screenshot, *Link* is activated for the function EXP (Expander), because both channel rows are no longer separated, but combined by bars. The functions OUTPUT and SETUP can not be deactivated nor set up separately, so no lines are shown at all. INPUT can't be deactivated, but in the stereo version of the DSP software the channels are linked.

To enter a sub menu first use the arrow keys to move the cursor (shown as a blinking of the corresponding three letters) in the desired direction. The Overview window will begin to scroll because not all functions can be displayed simultaneously. An arrow will show up at the left and right border of the display, indicating that some more functions are available (see picture above). The submenus are accessed by pressing *Enter*, a second press on *Enter* jumps back to the Overview window.

The button *Meter Mode* allows to switch the Peak Level Meter and the Correlator before (Input) or behind (Output) the DSP. This allows to check changes in level and phase caused by the activated functions without the need to switch the DSP in bypass mode.

Scrolling through the values with the +/- keys becomes accelerated when pressing the buttons for a longer time. A jump to the highest or lowest possible value is done by first pressing one, then the other key additional.

On the following pages you'll find detailed informations to all functions currently available in the factory shipped DSP software *Dual Mono A/D*.

7. General DSP Functions

7.1 Input

Function

Choice of input (Mic, Mic phantom powered, Line) per channel, invert phase (180°) per channel, activation of M/S processing.

Application

The setting *Mic. Phantom Powered* will apply 48 V DC to the XLR Mic jacks, so an operation with condenser microphones is possible without the need of an external supply.

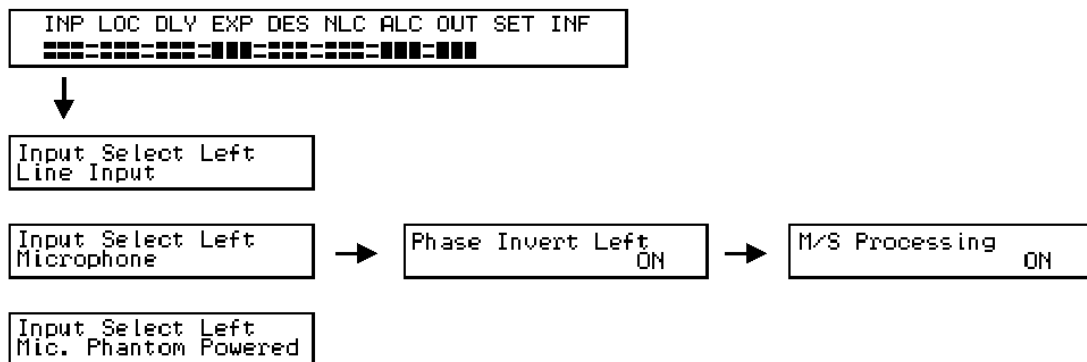
Inverting the phase allows to correct mics having a wrong polarity, or enables special effects in 'out of phase' technique.

The mid/side principle is a special positioning technique for microphones, which results in a mid signal on one channel and a side signal on the other channel. The M/S processing transforms these signals in a normal left/right stereo signal. Thus the mid/side signal is M/S decoded.

Settings

Place the cursor in the Overview window on **INP** and press Enter.

Menu



Options

Choice of Mic or Line input, activation and de-activation of phase invert and M/S processing.

Typical Settings

The mid/side processing automatically operates as encoder or decoder, depending on the signal feed. When processing a usual stereo signal all monaural informations will be shifted into the left channel, all stereo information into the right channel. Thus the stereo signal is M/S encoded. This gives some very interesting possibilities of manipulating the stereo base and generating stereo effects, as it is possible to process the stereo information with all the functions found in the ADI-96 PRO, like LOC, EXP, ALC and Delay. To avoid any processing of the left channel (the monaural information) simply set all parameters to their lowest value (so the functions won't change the audio data).

After this processing the signal must again run through the M/S processor to convert the mid/side signal back to a normal stereo signal. Many software audio editors (like Steinberg's WaveLab) include such an M/S function. That way, the second processing can be done in the digital domain, for example on a Windows PC.

7.2 Output

Function

Configuration of the signal at the digital output: Sample frequency, word length, Dither, word clock and Channel Status.

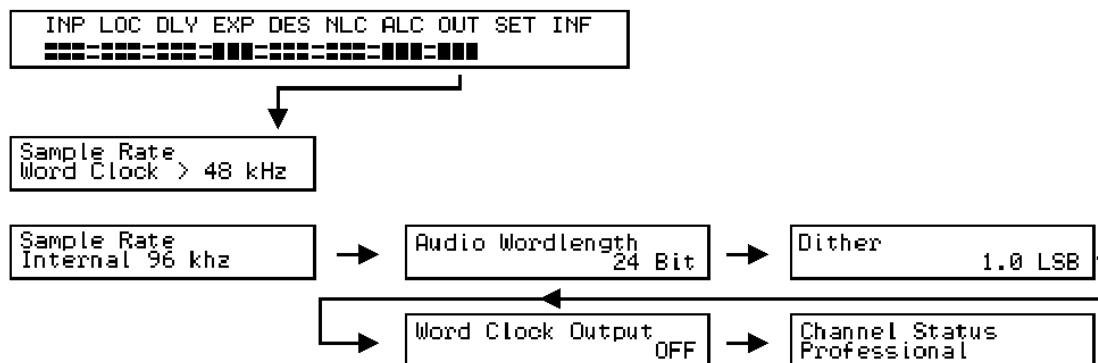
Application

Allows to set the parameters of the digitization, and allows to make the ADI-96 PRO fully compatible to all devices attached to its digital output.

Settings

Place the cursor in the Overview window on **OUT** and press Enter.

Menu



Options

Adjustable are:

- Sample rate 32 kHz, 44.1 kHz, 48 kHz, word clock < 64 kHz, 64 kHz, 88.2 kHz, 96 kHz, word clock > 48 kHz
- Audio word length 8 bit, 16 bit, 20 bit, 24 bit
- Dither OFF, 0.5 LSB, 1.0 LSB
- Word clock output OFF/ON
- Channel Status Professional, Consumer with copyright, Consumer without copyright

Typical Settings

In case the ADI-96 PRO shall be clocked from an external 44.1 kHz or 48 kHz word clock signal choose the setting *Word Clock < 64 kHz*.

Consumer devices like DAT and MiniDisc won't accept digital signals in Professional format at their SPDIF jacks. For this setup please choose *Consumer without Copyright*.

The choice of the word length determines the maximum resolution of the audio data at the ADI-96 PRO's output. At less than 24, for example when transferring on a 16 bit DAT, we recommend to choose *Audio Wordlength 16 Bit*, and also to activate the *Dither* function, to avoid distortions on very low level signals.

What Dither does can be easily shown at the setting *Audio Wordlength 8 Bit*, which results in a dynamic of only 48 dB at the output. Low level signals will suffer from high distortion. After activation of *Dither 1.0 LSB* a special noise signal is added which removes any audible distortion.

When activating *Word Clock Output ON* the device will provide a word clock signal of the current internal sample rate, available at the BNC jack on the rear of the device.

7.3 Setup

Function

Configuration of Peak Level Meter, LC-display and preset memory.

Application

Allows to change the display behaviour of all elements on the front panel, and to store and recall different configurations.

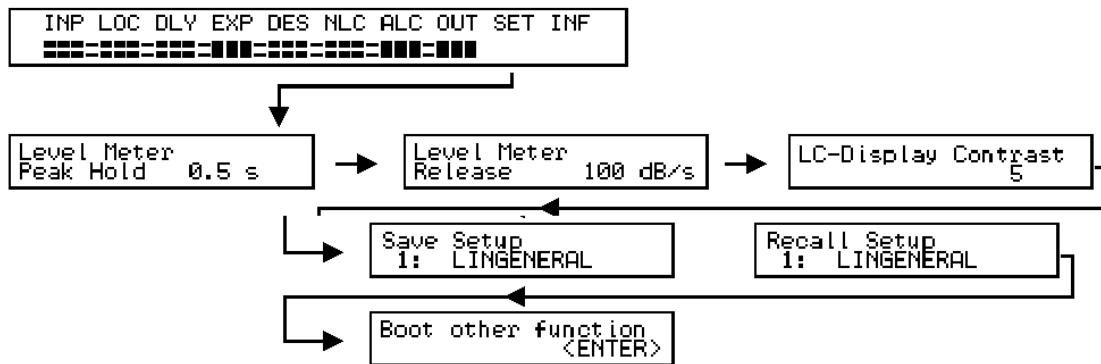
Dependent on the used DSP software the ADI-96 PRO includes up to 16 user presets. All settings except the ones found in the Setup menu will be stored (the setting of contrast and Level Meter is global). Each preset can be stored under an individual name. The desired number or letter is available by pressing the +/- keys, the arrow keys will move the cursor to the next position.

Copying one preset to another place can be done by first recalling the desired preset, than store it to the other location. A security function prevents overwriting data by mistake.

Settings

Place the cursor in the Overview window on **SET** and press Enter.

Menu



Options

Adjustable are:

- Level Meter Peak Hold 0.1 s up to 9.9 s
- Level Meter Release 1 dB/s up to 100 dB/s
- LC-Display Contrast 1 up to 10
- Save Setup 1 up to 16
- Recall Setup 1 up to 16
- Boot other function

Typical Settings

Most people will adjust the hold time of the peak level to around 1 second. A level meter with fast reaction is also desired by most people, for this the Level Meter Release should be set to the highest value of 100 dB/s.

Informations to the menu entry *Boot other function* can be found in the next chapter.

7.4 Boot other Function

Function

Starts the Bootloader.

Application

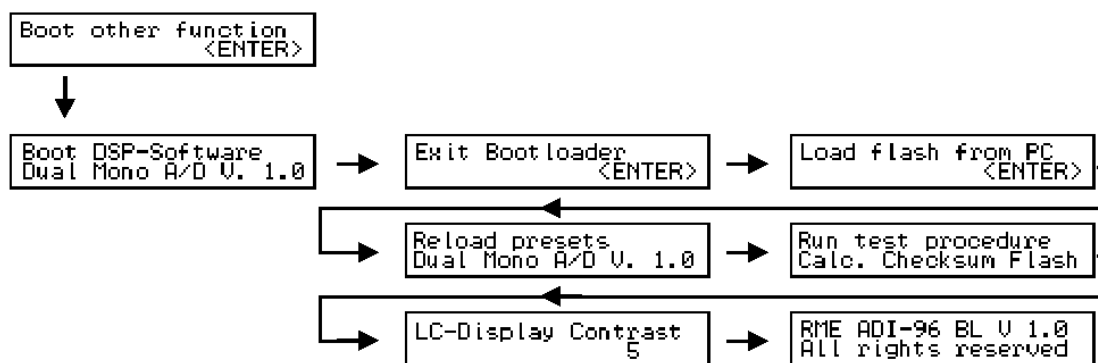
The Bootloader is nearly the same as the BIOS of a PC, it is the ADI's operating system at lowest level. Its main task is to load the DSP (audio) software after switching on the device. But there are some more functions it provides:

- Loading other DSP (audio) software from the internal memory
- Loading other DSP (audio) software from the serial interface
- Performing several hardware tests

Settings

Place the cursor in the Overview window on **SET** and press Enter, then use the right arrow key to scroll to *Boot other function*, then press Enter.

Menu



Options

Boot DSP-Software allows to choose from several DSP programs, stored in the ADI-96 PRO. Currently you'll find the DSP software *Dual Mono A/D*, an identical but easier to use version called *Stereo A/D*, and an *Audio Analyser* providing realtime 1/3 oct analysis. Updates will be available on our website for free and can easily be uploaded into the ADI-96 PRO using *Load Flash from PC* (see chapter 8. Serial Interface).

Reload Presets allows to reload all factory setups of the corresponding DSP software. This will erase all user settings.

The *Test Procedures* include the following tests:

- Calculate Checksum Flash (checks the Flash memory)
- Samplerate Generator (checks the function of the Clock PLL)
- Keyboard (allows to test all keys on the front panel)
- Display (lights up all LEDs and all signs of the LC-display)
- RAM Pattern Test (checks the RAM memory used by the DSP)

Typical Settings

None

7.5 Information

Function

Gives an overview on the current configuration. Shows the DSP software's version and name.

Application

The Information window gives a useful overview on which input is being used (Mic or Line), the active clock source (internal or word clock), the current sample frequency (shown in kHz) and word length. The ADI-96 PRO displays the Info window automatically after 100 seconds, in case no cursor key was pressed during this time.

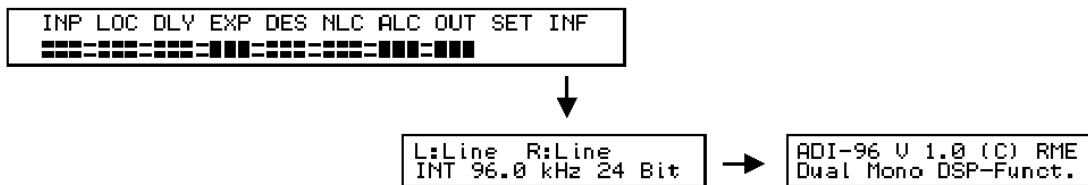
In word clock operation the Info window shows the actual frequency of the word clock signal feed, at missing word clock *WCK error* is shown.

The second Info window is automatically shown after switching on the device and shows name and version of the currently loaded DSP software.

Settings

Place the cursor in the Overview window on **INF** and press Enter.

Menu



Options

None

Typical Settings

None

7.6 Correlator

Function

Shows the correlation between left and right channel. Simply said: measures the phase between left and right.

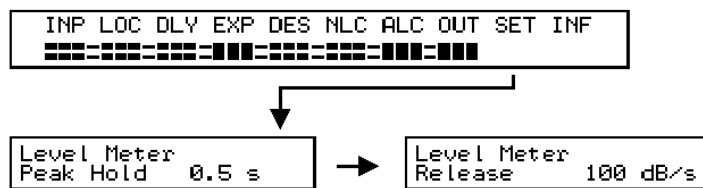
Application

Checking mono-compatibility, checking polarity of microphones, detecting sound problems due to delay and phase settings.

Settings

The Correlator has no settings, its behaviour depends on the release time set for the Peak Level Meter. Place the cursor in the Overview window on **SET** and press Enter.

Menu



Options

Level Meter Release 1 dB/s up to 100 dB/s.

Typical Settings

As with the Level Meter a faster setting (100 dB/s) is preferred by most users.

This function calculates the (mathematical) correlation between left and right channel. In simpler terms, the correlation gives an indication of the mono compatibility of the audio signal.

The value 1 measured with a sine corresponds to identical phase on both channels. In this case summing the channels (mono mode, reduced channel separation or a distant listening position) won't result in any loss of sound, the channels are 'in phase'. This is different with lower correlation values. In practice values between 1 and 0 indicate problem-free material.

0 corresponds to a phase deviation of 90° between left and right channel. An indication centered around 0 or slightly shifted into the red range (0 to -1) indicates material which might cause problems when converted to mono, or which might already suffer from frequency cancellations.

-1 means out of phase (180°). In this case a summing of the channels may result in a loss of sound, a stereo signal will completely erase itself when processed monaural.

The correlation meter can be used not only for the evaluation of existing material, but also for testing positions of microphones or their polarity before starting a recording.

8. Serial Interface

Function

Enables connection to and data transfer to and from any Windows PC.

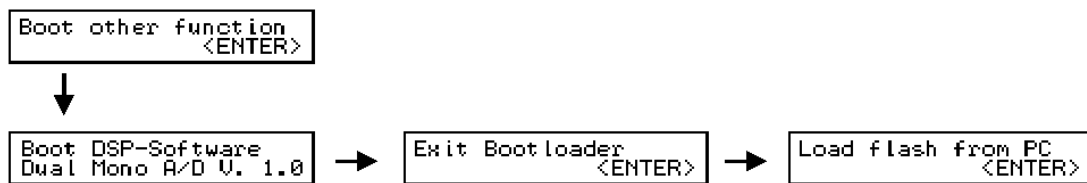
Application

Depending on the used software the ADI-96 PRO can be completely remote controlled, setups can be loaded/stored/edited, and new DSP software can be loaded into the device.

Settings


To load new DSP software: Place the cursor in the Overview window on **SET** and press Enter, then use the right arrow key to scroll to *Boot other function*, then press Enter, then scroll right to *Load flash from PC*.

Menu



Loading new DSP Software

Connect the COM-port of your Windows PC and the RS-232 interface of the ADI-96 PRO using the supplied D-type 9-pin cable. Start the Windows software **adsp_xx.exe** and choose the currently used port of the PC (COM1/COM2). Now the software waits for the ADI-96 PRO to start the transmission. Activate *Load Flash from PC* by pressing Enter at the ADI, the data transfer begins and ends automatically.


 *To activate the new software the ADI-96 PRO has to reboot with it. Change to the menu 'Boot DSP-Software' and use the +/- keys to choose the desired DSP software.*

Storing / Loading / Editing of Setups

These functions are available through the Windows software **Remote Control**, see chapter 9.9 Remote Control.

Reset

When transferring new DSP software over the serial interface into the ADI there is a slight chance the transmission might fail. In this case the ADI might be unable to boot and will hang when trying to load the incorrectly transferred software. To correct this, a new transfer or at least a reload of any internal software must be possible. This requires the Bootloader to be started. To ensure this the Bootloader is not only available through a menu of the DSP software, but also when switching on the device.

 *To activate the Bootloader press the Enter key while switching on the ADI-96 PRO. The Bootloader now allows to load internal software or a transfer via 'Load flash from PC'.*

9. DSP Software Dual Mono A/D

9.1 Brief Description and Characteristics

The DSP software *Dual Mono A/D* adds some outstanding audio functions to the ADI-96 PRO, required for a hi-class and convenient analog to digital conversion. Combined with the general functions of the ADI-96 PRO result in a versatile, supersonic Microphone/Line Audio Processor, replacing several external devices.

Dual Mono A/D includes the following functions:

Low Cut

High pass for reduction of low frequency noise

Delay

Delay of the audio signal per channel from 0 ms up to 170 ms, activation of Look Ahead

Expander

Dynamic level reduction to reduce noise in low level parts or between tracks

De-esser

Removes disturbing sibilants on speech and vocal recordings

Non Linear Compressor

Non linear compressor or Soft-Limiter. Similar to the effect known as tape saturation

Auto Level Control

Operates as Auto Leveler, Auto Gain and digital Compressor

Windows Software Remote Control

Complete remote control of the ADI-96 PRO through a PC. Visualisation of all external/internal levels. Storing/loading/editing of Setups

Look Ahead

RME's exclusive *Analog Processing* brings typical analog behaviour and sound to the digital domain. Our *Digital Processing* technology with Look Ahead minimizes changes in sound when using Expander and Auto Level Control.

9.2 Low Cut

Function

Hi-pass with adjustable corner frequency between 2 Hz and 250 Hz at 12 dB/oct.

Application

Removes low frequency noise in microphone recordings, rumble and flutter when converting older analog material, hum in speech recordings.

Settings

In the Overview window place the cursor on **LOC** and press Enter.

Menu



Options

The begin of the bass suppression (corner frequency) is adjustable in each channel from 2 Hz to 250 Hz.

Typical Settings

In mixing desks values around 120 Hz are usual. As the filter has a smooth roll-off 50 Hz hum will be noticeably reduced with this setting, but the harmonics at 100 Hz and above won't. That's why the filter can work up to 250 Hz. In extreme situations such a setting is able to rescue a speech recording.

Noise and the low frequency pops of microphones do not require such an extreme cutoff at lower frequencies. Recommended values are 50 to 80 Hz.

Low frequency rumble as of turntables (or other signals which lead to slowly pumping loudspeakers) are already suppressed with a setting of 30 Hz, which results in no audible bass reduction. When mastering often all frequencies below 30 Hz are lowered. Several electronic sound sources, especially Synthesizers and Keyboards, generate high amounts of inaudible low frequency junk, which stresses power amps and loudspeakers, but has no effect on the feelable and audible bass bottom.

9.3 Delay

Function

Delay of the audio signal per channel from 0 ms up to 170 ms, activation of Look Ahead.

Application

Allows to correct the time alignment of left and right channel, for example when doing microphone recordings. Allows to correct the time alignment of PA systems. Allows to produce stereo effects by delaying one channel. Activates the Look Ahead mode for optimized sound quality with Expander and Auto Level Control.

Settings

Place the cursor in the Overview window on **DLY** and press Enter.

Menu



Options

The delay of the audio signal can be set from 0 ms up to 170 ms per channel.

Typical Settings

When doing a stereo microphone recording where the mics don't have an identical distance to the sound source the Delay function allows to delay one channel so the signal at the digital output of the ADI-96 PRO is time aligned again.

The speed of sound is about 340 m/s. A difference of one meter in the above situation already requires a correction by 3 ms! The Correlation meter (see chapter 7.6 Correlator) is a valuable help in fine tuning the necessary correction, but for the coarse values you'll also need a calculator and a foot rule.

Other time alignment problems may require to delay both channels in an identical manner. The maximum setting of 170 ms enables a simultaneous output of loudspeakers in a distance of 56 meters together with loudspeakers placed near the listening person.

The human ear will detect a signal as monaural when amplitude (loudness) and time related output of left and right channel are identical. When one channel is delayed up to 10 ms the signal which formerly seemed to come out of the middle will fall apart into two signals, individually sent by left and right channel. This may be used as some kind of simple stereo effect generator, but a very good generator needs some more tricks like modulation on the delay, short reverb and changes in the frequency response, which are not available here.

One way to achieve a very efficient stereo manipulation is to use the M/S processing (encoding), as the stereo informations on the right channel can be single processed by Compressor and Delay within the ADI. See chapter 7.1.

Expander and ALC will enter the Look Ahead mode in case the Delay is activated and the delay time is at least 0.1 ms. Informations on how to set up the Delay for best performance with Expander and Auto Level Control can be found in chapter 9.8.

9.4 Expander

Function

Level dependent level reduction, each channel with adjustable threshold (-30 dB to -110 dB), release (0.1 s to 25.5 s), ratio (1.2 to 5.0) and attack time (Fast 1 ms or Slow 25 ms).

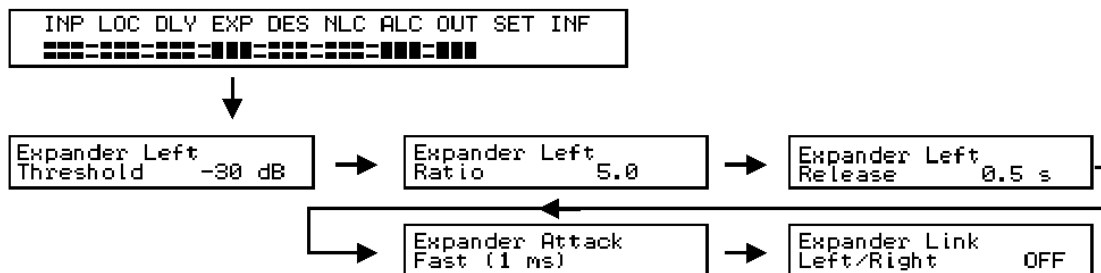
Application

Allows broadband suppression of noise, removal of noise and hum in low level parts or between tracks, and de-compression of highly compressed material.

Settings

Place the cursor in the Overview window on **EXP** and press Enter.

Menu



Options

Adjustable are:

- Threshold -30 dB to -110 dB. When the overall level is lower than this value the Expander begins to reduce the level.
- Ratio 1.2 to 5.0. Defines the ratio between original and reduced level.
- Release 0.1 s to 25.5 s. Defines the time needed to reach 20 dB level reduction after the last overall level exceeded the threshold value.
- Attack Fast (1 ms) or Slow (25 ms).
- Link Left/Right. Avoids pan shifts by combined controlling of both channels.

Typical Settings

As opposed to a Noisegate an Expander does not switch off audio when the Threshold is no longer reached. Instead the level is lowered smoothly according to the chosen ratio. A ratio of 1.2:1 equals nearly no change, because a change in input level of 10 dB leads to a 12 dB change in output level (only 2 dB difference). Typical Ratio values are 2 to 3 at a Release of 2 s. That way, music can be faded in and out without heavy side effects. Noise between tracks can be reduced easily.

The Threshold of the Expander has a strong relationship with the audio material and the noise in it. LPs often require a Threshold of -30 dB (plus Attack Slow, see chapter 9.8), while recordings from hi-quality sources often only need values around -75 dB. At values of -100 dB the dynamic range of the ADI-96 PRO can be increased.

With more extreme settings like Ratio 5:1 and Release 0.1 s the Expander operates like a Noisegate. With these settings it is possible for example to isolate percussive instruments from environmental noise.

The settings Ratio 1.5:1 and a Release of around 0.5 s allows to de-compress signals which were too strong compressed before. This will result in a more natural, open sound, but the original can't be recovered completely.

9.5 De-esser

Function

Selective dynamic level limiter with automated relative level control. Removes disturbing sibilants on speech and vocal recordings, or other level peaks present in its operating range.

Application

In general there are several methods used to reduce sibilants:

Fixed bandpass - an Equalizer with fixed cut in the sibilant frequency area. Results in a constant muffled sound. Not useable on masters.

Dynamic lowpass - cuts all high frequencies according to the level found in the sibilant area. Requires a threshold setting adopted to the actual signal. Not useable on masters.

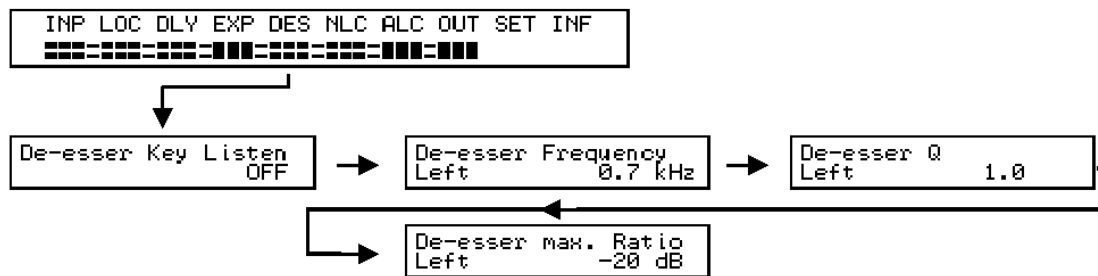
Dynamic bandpass - Requires a threshold setting adopted to the actual signal.

The method used in the ADI-96 PRO is a dynamic bandpass with all its advantages, but does no longer need a threshold to be set according to the actual signal. Using an automated relative function the overall level is compared to the level in the sibilant area, the threshold is automatically set to always achieve the same sound. The level of the actual signal doesn't matter anymore.

Settings

Place the cursor in the Overview window on **DES** and press Enter.

Menu



Options

Adjustable are:

- Key Listen. Allows to monitor the filtered signal. Helps to find the best filter setting for an ideal signal reduction within the desired frequency area
- Frequency 0.7 kHz up to 20 kHz. Center frequency of the bandpass
- Q 0.2 up to 10. Quality factor, defines the width of the bandpass (width of controlled area)
- Ratio 0 dB up to -50 dB. Relative level reduction of the bandpass area

Typical Settings

Sibilants will show up between 4 kHz and 8 kHz. Due to the nearly inaudible operation of the ADI-96 PRO the Q parameter can be set to a lower value (thus resulting in a larger de-essed area), for example 1. The Ratio setting depends on the desired sound and the previously applied equalizing. In an already mastered material a Ratio of up to -10 dB is possible without getting noticeable reductions. Pure speech and vocals can be processed at settings of up to -30 dB. Too high a value will result in a constant treble reduction, although the De-esser is still working as a dynamic filter.

Because of the automated relative level control the setting once found to meet your personal taste does not need any further changes even when the input level is totally different.

9.6 Non Linear Compressor

Function

Non linear compressor. The input signal is processed with a special amplitude function which softly cuts the highest level peaks. The resulting third order harmonics are widely masked by the original signal, thus inaudible in most situations. Similar to the effect known as tape saturation.

Application

Allows to provide additional headroom of up to 5 dB without audible clipping (clipping: brickwall limiting leading to high distortion). Allows to increase the loudness without increasing the maximum peak level (similar to the effect known as Maximizer).

Settings

Place the cursor in the Overview window on **NLC** and press Enter.

Menu



Options

Adjustment of the desired gain in loudness (level) from 1 dB to 5 dB.

Typical Settings

As long as not very delicate signals from solo instruments or complete mixes are processed the *Non Linear Compressor* is a marvellous tool for every recording and may be always activated at *Gain 5 dB*. It helps especially when doing live acoustical recordings to gain additional headroom and operates like a compressor without attack or release time, thus operating virtually inaudibly.

The NLC may be used to achieve increased loudness without any pumping when processing material already normalized near 0 dBFS. When used in combination with the ALC and fast settings the increase in loudness can be more than extreme, but will also result in compacted death material without any dynamic left.

9.7 Auto Level Control

Function

Automatic level control with adjustable gain range and adjustable operating speed per channel.

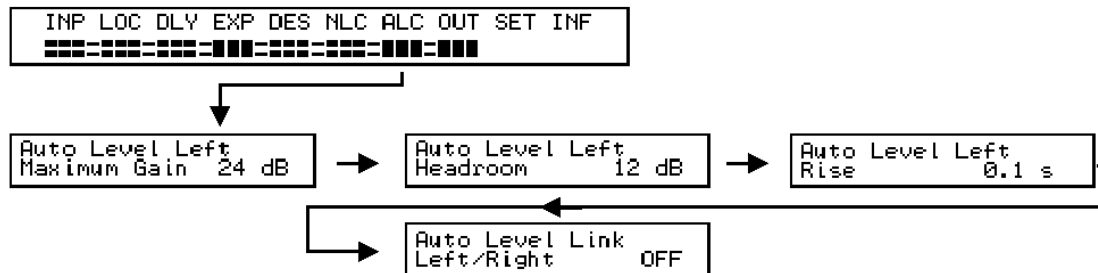
Application

Serves as automatic level control for long term recordings, raises the volume to avoid low level recordings. Useable as Leveler or - at faster settings - digital Compressor.

Settings

Place the cursor in the Overview window on **ALC** and press Enter.

Menu



Options

Adjustable are:

- Maximum Gain 0 up to 24 dB. Defines the maximum gain in level.
- Headroom 1 up to 12 dB. Ratio between average level and maximum (0 dBFS) level.
- Rise 0.1 s up to 25.5 s. Time needed for a 20 dB gain change.
- Link Left/Right. Avoids pan shifts by combined controlling of both channels.

Typical Settings

A usual compressor reduces the dynamic range *above* a defined level (Threshold) using a defined Ratio (for example 3:1). The ALC raises all signals *beneath* the maximum level (0 dBFS) until they reach 0 dBFS minus *Headroom*. The maximum level gain is 24 dB, so a signal at -30 dBFS will be raised to a level of -6 dBFS. Because of the finite reaction time the average amplification has to be limited by the parameter *Headroom* to a value beneath 0 dBFS. In dynamic material Headroom determines the ratio between peaks (maximum 0 dBFS) and average level.

When doing acoustic recordings the ALC can improve the average level significantly, recommended settings are up to 15 dB Gain. Recordings from uncritical sound sources (like speech) can be processed up to 24 dB, so the ALC serves as a real automatic level control. The Rise time should be longer than 10 s to avoid unwanted pumping. The ALC will change the level slowly and smoothly in a non-disturbing way.

At very short Rise settings the ALC operates as digital compressor. The parameter *Rise* then equals the commonly known *Release*. Most important is the parameter *Headroom*. A setting of 1 dB increases the loudness, but will also lead to a 3 ms cutoff of all peaks (Clipping). Because of the short time this is normally inaudible. In case the ALC shall preserve a more natural sound settings of more than 3 dB are recommended. From 7 dB up to 12 dB the internal attack time is doubled to 6 ms, because the higher headroom results in very rare clipping. After activation of a sufficient delay time (see chapter 6.3 Delay) the ALC works in Look Ahead mode without any clipping (see chapter 9.8).

The combination of NLC set to Gain 5 dB and ALC set to Headroom 1 dB, Gain 5 dB and Rise 2 s results in an extreme increase of loudness at very little side effects (pumping), similar to the effect known as *Maximizer*.

9.8 Look Ahead

9.81 Expander

When set to *Fast* the Expander has a very fast attack time of less than 1 ms. However such a low reaction time already leads to a smoother sound when processing percussive signals with strong attack noises (click). The smoothing depends on the actual Gain Reduction. With high Gain Reduction the Expander needs a longer time to open (between 0.1 and 1.5 ms). A faster reaction of the Expander is not possible, as it will cause clicks and pops.

The damping on the beginning of the signal can easily be avoided by delaying the audio signal, but not the control signal. With this the Expander will open although the audio signal isn't present yet. This method of processing with a preview is called *Look Ahead*.

The ADI-96 PRO has no extra switch to activate the Look Ahead mode. As Look Ahead requires a delay of the audio signal we found it both logical and easy to use when Look Ahead is simply activated by the already implemented function Delay. As soon as the Delay is activated and the delay time is at least 0,1 ms the Expander will work in Look Ahead mode. Please note that the above problem can only be addressed by a minimum delay time of at least 1,5 ms. Whether the chosen delay time is sufficient or not can easily be checked by using the function *Bypass DSP*.

When set to *Slow* the attack time is 25 ms. This is necessary to avoid a reaction (an opening) on very short noises like cracks and clicks, for example when digitizing LP's. The long attack time can again easily be compensated by a delay of 25 ms, so there is no loss of sound on material which starts without any fade in.

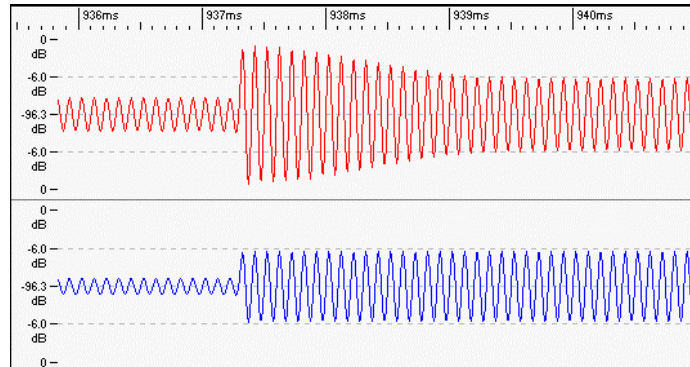


The usage of Look Ahead leads to a delay of the digitized signal, which might be a serious problem in recording situations where realtime behaviour is required. Please note that the function EXP was optimized to give excellent results even without Look Ahead.

9.82 ALC

The ALC operates with two different attack times. Headroom 1 to 6 dB uses an attack time of 3 ms, Headroom 7 to 12 dB of 6 ms. The picture printed below shows the internal operation of the ALC. The left channel (top) is amplified by the defined Gain (in this case 10 dB), as the defined average level (0 dBFS minus Headroom) isn't reached. The right channel (bottom) shows the unprocessed signal.

A higher input level at this gain setting will either exceed the average level (0 dBFS minus Headroom) or even the maximum level of 0 dBFS. Then the ALC immediately begins to reduce the amplification to avoid overloads. After 3 ms (or earlier) the signal level is back to a correct state. Exceeding the maximum level in this short time will cause precise clipping.



Because of the short time this clipping occurs there will normally be no noticeable side effects. In case the input level drops the ALC will begin to raise the level, with a speed according to the time set under *Rise*. This technology results in an extraordinary digital Compressor. The precise and controlled clipping leads to an analog behaviour and 'punchy' sound characteristic.

A level control without any clipping can be achieved by using the ALC in *Look Ahead* mode. As with the Expander activating the Delay and setting a sufficient delay time will avoid any clipping and lead to an extreme clean processing. Usual settings between 5 and 10 dB Gain need only 4 ms to prevent clipping. Very high Gain settings (up to 24 dB) require greater preview times. 25 ms is enough for secure operation at any possible setting.

The sound difference between *Look Ahead* and normal operation can easily be checked in realtime by simply switching the Delay on and off.



The usage of Look Ahead leads to a delay of the digitized signal, which might be a serious problem in recording situations where realtime behaviour is required. Please note that the function ALC was optimized to give excellent results even without Look Ahead.

The preview time of *Look Ahead* is internally limited to 25 ms. Higher delay times will not result in a longer preview time. This guarantees a correct correlation between audio and control signal.

9.9 Remote Control

Function

Complete remote control of the ADI-96 PRO through a Windows PC. Visualisation of all external/internal levels. Storing/loading/editing of setups.

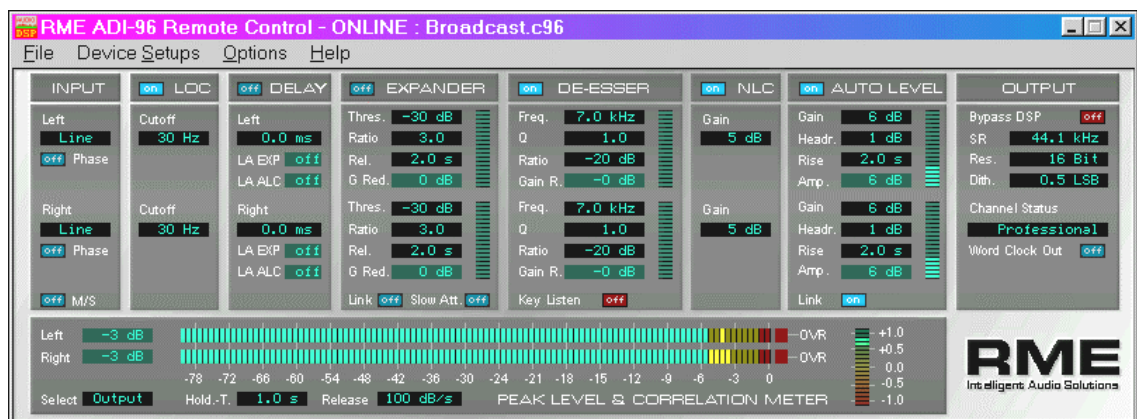
Application

Connect the COM-port of your Windows PC and the RS-232 interface of the ADI-96 PRO using the supplied D-type 9-pin cable. Start the Windows software **adipro.exe** and choose the currently used port of the PC (COM1/COM2). The ADI-96 PRO requires no further settings, requests of the software *Remote Control* via the serial interface are automatically detected.

Operation/Settings

After starting the software the ADI can be set by both the PC software or the keys on the device itself. Any change at the device is shown immediately in the software, all changes in the software are immediately shown on the ADI's display.

Each function is activated by a single mouse click on the corresponding ON/OFF button. Changing the parameters is done by a double click onto any parameter, which brings up a window to enter new values using the keyboard. Another method is to keep the left mouse button pressed when moving up or down from any parameter field. The corresponding value will change according to the movement of the mouse. Press the shift key while moving the mouse to change the values in bigger steps.



The picture shows the Windows software Remote Control in *Dual Mono A/D* mode. The program includes an extensive online help which describes all functions and elements of the user interface. Most interesting and helpful is the visualisation of the 'control voltages' of Expander, De-esser and Auto Level Control. At Expander and De-esser two LED chains indicate the actual Gain Reduction. The LED chains of the ALC indicate the amount of amplification. The LED chains are a valuable help when setting up the functions. Their constantly updated states are also shown as numerical values in the corresponding fields.

The display of the actual Look Ahead state found in the Delay section depends on ALC, EXP and Delay. Look Ahead is already active at 0.1 ms delay time, but has no effect at such a small value. Therefore LA EXP ON is shown at 1 ms, LA ALC at 4 ms or above.

At the bottom of the software the input and output level is displayed according to the setting 'Meter Mode'. This meter displays a level range of 80 dB in steps of 1 dB.

Storing / Loading / Editing of Setups

The software provides the following functions: reading setups from the ADI-96 PRO, saving setups on the PC's storage media, loading setups from PC into ADI-96 PRO.

9.10 Table of Parameters (Presets Dual Mono A/D)

Setup Name	LINE GENERAL	MIC GENERAL	ACOUSTIC	VOCAL	SPEECH	LINE GATE	MIC GATE	LINE COMP	MIC COMP
Phase invert Left	off	off	off	off	off	off	off	off	off
Phase invert right	off	off	off	off	off	off	off	off	off
M/S-Processing	off	off	off	off	off	off	off	off	off
LOC	on	on	on	on	on	off	on	on	on
EXP	off	off	off	on	on	on	on	on	on
Expander link	off	off	off	on	off	on	on	on	off
NLC	off	off	off	on	on	off	on	on	on
DLY	off	off	off	off	off	off	off	off	off
DES	off	off	off	on	on	off	off	off	off
De-esser Key listen	off	off	off	off	off	off	off	off	off
ALC	on	on	on	on	on	off	off	on	on
Autolevel link	on	on	on	on	on	on	on	on	on
Wordclock Output	off	off	off	off	off	off	off	off	off
Input Select Left	LINE	MIC	MIC	MIC	MIC	LINE	MIC	LINE	MIC
Input Select Right	LINE	MIC	MIC	MIC	MIC	LINE	MIC	LINE	MIC
Lowcut Freq. left	2	40	50	80	120	2	100	30	80
Lowcut Freq. right	2	40	50	80	120	2	100	30	80
Exp. Attack	Fast	Fast	Fast	Fast	Fast	Fast	Fast	Fast	Fast
Exp. Thres. left	-100	-100	-90	-40	-50	-70	-40	-70	-40
Exp. Ratio left	2	2	2	3	3	5	5	2	2
Exp. Release left	5	5	5	2	2	0,1	0,1	2	2
Exp. Thres. right	-100	-100	-90	-40	-50	-70	-40	-70	-40
Exp. Ratio right	2	2	2	3	3	5	5	2	2
Exp. Release right	5	5	5	2	2	0,1	0,1	2	2
NLC left	5	5	5	5	5	5	5	5	5
NLC right	5	5	5	5	5	5	5	5	5
Delay left	0	0	0	0	0	0	0	0	0
Delay right	0	0	0	0	0	0	0	0	0
Dees. Freq. left	7	7	7	7	7	7	7	7	7
Dees. Q left	1	1	1	1	1	1	1	1	1
Dees. Ratio left	-20	-20	0	-20	-20	0	-20	0	-20
Dees. Freq. right	7	7	7	7	7	7	7	7	7
Dees. Q right	1	1	1	1	1	1	1	1	1
Dees. Ratio right	-20	-20	0	-20	-20	0	-20	0	-20
ALC max. Gain left	5	5	5	15	20	0	6	10	15
ALC Headroom left	-6	-6	-6	-6	-6	-6	-3	-6	-3
ALC Rise left	5	5	5	2	5	20	2	2	2
ALC max. Gain right	5	5	5	15	20	0	6	10	15
ALC Headroom right	-6	-6	-6	-6	-6	-6	-3	-6	-3
ALC Rise right	5	5	5	2	5	20	2	2	2
Channel Status	CON	CON	PRO	PRO	PRO	PRO	PRO	PRO	PRO
Sample Rate	44.1kHz	44.1kHz	96kHz	96kHz	44.1kHz	96kHz	96kHz	96kHz	96kHz
Wordlength	16Bit	16Bit	24Bit	24Bit	24Bit	24Bit	24Bit	24Bit	24Bit
Dither	0.5LSB	0.5LSB	0.5LSB	0.5LSB	0LSB	0LSB	0LSB	0.5LSB	0.5LSB

Setup Name	MAXIMIZER	LP	CASSETTE	LINE BYPASS	MIC BYPASS	USER	USER
Phase invert Left	off	off	off	off	off	off	off
Phase invert right	off	off	off	off	off	off	off
M/S-Processing	off	off	off	off	off	off	off
LOC	on	on	on	off	off	off	off
EXP	off	on	on	off	off	off	off
Expander link	off	on	on	off	off	off	off
NLC	on	on	off	off	off	off	off
DLY	off	on	on	off	off	off	off
DES	off	off	off	off	off	off	off
De-esser Key listen	off	off	off	off	off	off	off
ALC	on	off	off	off	off	off	off
Autolevel link	on	on	on	off	off	off	off
Wordclock Output	off	off	off	off	off	off	off
Input Select Left	LINE	LINE	LINE	LINE	MIC	LINE	LINE
Input Select Right	LINE	LINE	LINE	LINE	MIC	LINE	LINE
Lowcut Freq. left	30	40	30	2	2	2	2
Lowcut Freq. right	30	40	30	2	2	2	2
Exp. Attack	Fast	Slow	Slow	Fast	Fast	Fast	Fast
Exp. Thres. left	-100	-30	-30	-110	-110	-100	-100
Exp. ratio left	2	2,5	2,5	2	2	2	2
Exp. release left	5	2	5	5	5	5	5
Exp. Thres. right	-100	-30	-30	-110	-110	-100	-100
Exp. Ratio right	2	2,5	2,5	2	2	2	2
Exp. Release right	5	2	5	5	5	5	5
NLC left	5	5	5	5	5	5	5
NLC right	5	5	5	5	5	5	5
Delay left	0	25	25	0	0	1	0
Delay right	0	25	25	0	0	1	0
Dees. Freq. left	7	7	7	7	7	7	7
Dees. Q left	1	1	1	1	1	1	1
Dees. Ratio left	0	0	0	0	0	0	0
Dees. Freq. right	7	7	7	7	7	7	7
Dees. Q right	1	1	1	1	1	1	1
Dees. Ratio right	0	0	0	0	0	0	0
ALC max. Gain left	5	3	3	0	24	0	0
ALC Headroom left	-1	-6	-6	-6	-6	-6	-6
ALC Rise left	2	2	10	25	25	25	25
ALC max. Gain right	5	3	3	0	18	0	0
ALC Headroom right	-1	-6	-6	-6	-6	-6	-6
ALC Rise right	2	2	10	25	25	25	25
Channel Status	CON	CON	CON	PRO	PRO	CON	CON
Sample-Rate	44.1kHz	44.1kHz	44.1kHz	96kHz	96kHz	48kHz	44.1kHz
Wordlength	16Bit	16Bit	16Bit	24Bit	24Bit	24Bit	24Bit
Dither	0LSB	0LSB	0LSB	0LSB	0LSB	0.5LSB	0.5LSB

10. DSP-Software Stereo A/D

10.1 Brief Description and Characteristics

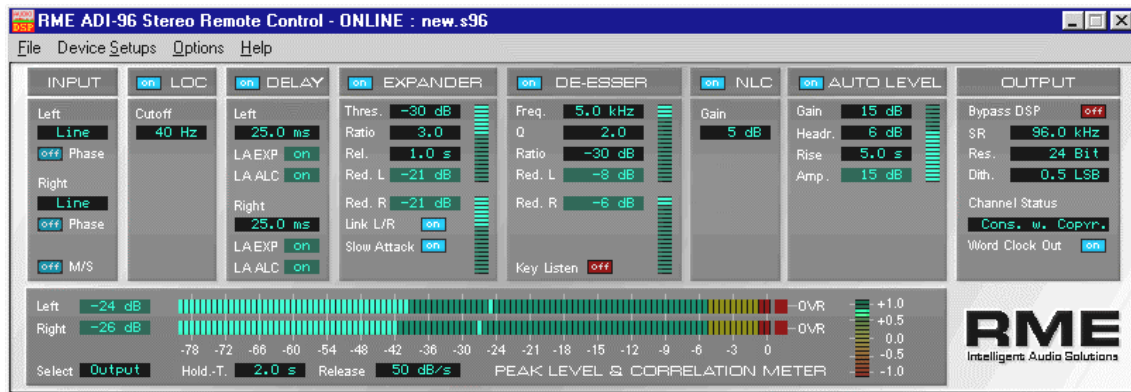
The DSP software *Stereo A/D* is an easier to configure and to handle version of *Dual Mono A/D*. The single setting of each channel was - as far as possible - reduced to a common setting of both channels. This is sufficient for most applications. The available DSP functions are exactly the same as in *Dual Mono A/D*, and are explained in detail in chapter 9.

To avoid shifts between left and right normally all functions should operate in a link mode (both channels always react identical). This is default in *Stereo A/D*. Please note that the De-esser does not work linked at all. As the Expander might be used as noise gate, and suffer from one channel signals, Link can be deactivated.

The number of user definable setups in mode *Stereo A/D* is limited to six.

10.2 Remote Control

The software Remote Control offers to control all functions and parameters of *Stereo A/D* mode through a Windows PC. This includes a visualisation of all external and internal levels and the storing/loading/editing of setups. Detailed informations can be found in chapter 9.9.



The picture shows the Windows software Remote Control in *Stereo A/D* mode.

10.3 Table of Parameters (Presets Stereo A/D)

Setup-Name	LINE GENERAL	MIC GENERAL	LINE COMP	MIC COMP	MAXIMIZER	LP
Phase invert Left	off	off	off	off	off	off
Phase invert right	off	off	off	off	off	off
M/S-Processing	off	off	off	off	off	off
LOC	on	on	on	on	on	on
EXP	off	off	on	on	off	on
Expander link	off	off	on	off	off	on
NLC	off	off	on	on	on	on
DLY	off	off	off	off	off	on
DES	off	off	off	off	off	off
De-esser Key listen	off	off	off	off	off	off
ALC	on	on	on	on	on	off
Wordclock Output	off	off	off	off	off	off
Input Select Left	LINE	MIC	LINE	MIC	LINE	LINE
Input Select Right	LINE	MIC	LINE	MIC	LINE	LINE
Lowcut Freq. left	2	40	30	80	30	40
Exp. Attack	Fast	Fast	Fast	Fast	Fast	Slow
Exp. Thres. left	-100	-100	-70	-40	-100	-30
Exp. Ratio left	2	2	2	2	2	2,5
Exp. Release left	5	5	2	2	5	2
NLC left	5	5	5	5	5	5
Delay left	0	0	0	0	0	25
Delay right	0	0	0	0	0	25
Dees. Freq. left	7	7	7	7	7	7
Dees. Q left	1	1	1	1	1	1
Dees. Ratio left	-20	-20	0	-20	0	0
ALC max. Gain left	5	5	10	15	5	3
ALC Headroom left	-6	-6	-6	-3	-1	-6
ALC Rise left	5	5	2	2	2	2
Channel Status	CON	CON	PRO	PRO	CON	CON
Sample Rate	44.1kHz	44.1kHz	96kHz	96kHz	44.1kHz	44.1kHz
Wordlength	16Bit	16Bit	24Bit	24Bit	16Bit	16Bit
Dither	0.5LSB	0.5LSB	0.5LSB	0.5LSB	0LSB	0LSB

11. DSP Software Audio Analyser

11.1 Brief Description and Characteristics

The DSP software *Audio Analyser* transforms the ADI-96 PRO into a powerful Spectrum Analyser. As opposed to nearly all current PC based solutions no FFT (Fast Fourier Transform) is used. Although this method needs only little calculation (CPU) power, it offers no useable translation of sound in vision. The 'bands' of a FFT have a constant frequency distance, which results in the higher area in numerous bands, but in the lower frequencies in only some bands. Some manufacturers try to minimize this behaviour, but even then a sine will show an un-symmetrical display and an insufficient band separation.

The DSP of the ADI-96 PRO performs a real bandpass filter calculation, as usual in all professional 'Audio Spectrum Analyzers'. The distance between the filters is not linear, but scaled according to human hearing. Thanks to the external calculations the Windows software does only display the results. This allows to run a 30 band Analyser with 50 dB range, sharp filters and 100 LED's per band even on an older PC, without driving the computer to its limits. On a Pentium III 800 the CPU load is around 5%.

Thanks to its complete digital operation the *Audio Analyser* offers features previous unavailable from analog Analyzers. These features include freely adjustable attack and release time, a display configurable in many ways, different filters and special display modes. Because of the hi-quality AD-conversion the ADI-96 PRO offers its *Audio Analyser* will precisely show levels down to -120 dB.



Please note that the ADI-96 PRO will even in Audio Analyser mode operate as high end Mic/Line AD-converter! But the functions provided by the 'Dual Mono A/D' software (Low Cut, Delay, Expander, De-esser, NLC, ALC) are not available in Analyser mode.

11.2 Hotkeys

Some functions are directly accessible using predefined shortcuts:

Left	L
Right	R
L+R (Mono)	M
Freeze	space
RMS/Peak Level	D

Furthermore there's direct access to the following menu entries:

Help - Contents	F1
Analyser Settings	F2

The keys F5 to F8 offer a fast switch function between four different setups. The setups can be changed anytime, as they consist of the four (supplied) files **fastkey5.a96** to **fastkey8.a96**. Load the according setup using the menu's entry *File/Open Setup File*, change the settings to your needs. Then store the settings using *Save Setup File*, but don't change the name! Those four files must reside in the same directory as the main program **adipro.exe**!

11.3 Software Audio Analyser

Function

Visualization of the audio signal according to level and frequency.

Application

Connect the COM-port of your Windows PC and the RS-232 interface of the ADI-96 PRO using the supplied D-type 9-pin cable. Use *Boot other function* at the ADI-96 PRO to boot the DSP software *Audio Analyser*. Another way is to start the Windows software **adipro.exe** and choose *Boot other function/Audio Analyser* in the *Options* menu of the software. Don't forget to check the currently used port of the PC (COM1/COM2).

Operation/Settings

The visualisation on the screen is not mainly set by the ADI-96 PRO, but by the Windows software. That's why only some parameters are directly configurable at the device. After starting the software use F2 or the right mouse key to open the window *Analyser Settings*. Here all parameters can be adjusted in a convenient way. The guidelines of using *Remote Control* (see chapter 9.9) are the same for this software.



Analyser Settings

The window *Analyser Settings* includes four function areas:

INPUT

Choice of input (Mic, Mic phantom powered, Line) per channel, invert phase (180°) per channel, activation of M/S processing. See chapter 7.1.

SPECTRUM ANALYSER

Holds all parameters controlling the display of the audio signal in the analyser window. Additional the display modes are set. See chapter 10.4.

LEVEL M.

Activation of display and configuration of the the RMS/Peak Level meter: Peak Hold Time 0.1 s to 9.9 s, Release 1 dB/s to 100 dB/s

OUTPUT

Configuration of the signal at the digital output: Sample frequency, word length, Dither, word clock and Channel Status. See chapter 7.2.

Storing / Loading / Editing of Setups

The software allows to store, load and edit setups from and to the PC's storage media. No setups will be stored to or loaded from the ADI-96 PRO.

11.4 Settings Spectrum Analyser

Type

Number and Quality factor of displayed bands. Available are:

- 10 band (1/1 oct. steps), Quality factor 1.4 or 2.5
- 15 band (2/3 oct. steps), Quality factor 1.4 or 3
- 20 band (1/2 oct. steps), Quality factor 3 or 5
- 30 band (1/3 oct. steps), Quality factor 3 or 5

The width of the display window is automatically adjusted to the number of of bands. The values printed on the x-axis are rounded for better readability. At a sample rate of 32 kHz the band 20 kHz isn't available, as no frequencies above 16 kHz are sampled. The choice of Quality factor defines the crosstalk between the bands.

Source

Sets the channel to be displayed. Adjustable to left, right, left plus right (monaural) and left minus right (only stereo).

Rise Time

Rise time of the display, adjustable from 0 ms to 200 ms. Values below 5 ms lead to a display even of shortest peaks. Values around 7 ms will lead to a display which behaves more like the audio heard. Values greater than 30 ms result in a display similar to a VU meter.

Release

Release time of the display, referenced to 20 dB. Adjustable from 0,3 s to 20 s. A good readable display is achieved at around 2 seconds. When measuring using noise signals 20 seconds result in a smooth and well readable display.

Hold Time

Hold time of the optional peak level meter (mode *Peak Hold*), adjustable from 0.1 s to 9.9 s.

Range

Visible display range. Choose between 30 dB and 50 dB.

Top Level

Highest level displayed in the analyser window. Serves to adapt the display to the actual signal level. Available settings are 0 up to -70 dB in steps of 10 dB.

Mode

Kind of display. *Bar* shows a solid bar, *Point* only one LED/segment per band. *Peak Hold* combines bar and peak display. The peak portion is defined by *Hold Time*.

Size

Size of the analyser window. Choose between small, medium and big.

Resolution

Range in dB per LED/segment. Choose between 0.5 dB, 1 dB and 2 dB.

Instant Hold

Sets the release time in *Bar* and *Point* mode to infinite. In mode *Peak Hold* only the peak display will be set to infinite release.

Reset

Resets all peak values covered by *Hold Time* or *Instant Hold*.

Freeze

Freezes the display to the actual state. Snap shot function.

11.5 Notes on Operation

The most important application using a Spectrum Analyser is the visualisation of frequencies and levels found in music or speech. What you see is what you hear! The Analyser shows levels and frequencies even at the edge of the human ear's abilities. The visual display helps to train your ears, and avoids serious mistakes when mixing to the master tape. Usual studio monitors won't let you hear frequencies below 100 Hz. Simply look on the *Audio Analyser* to see what's going on in the underground!

Reading the display is not easy for novices because of the huge amount of information it shows. But after some training you'll agree that this tool is a precious help in every day work.

Analysis of Music

Attack and Release time are the most important parameters when trying to achieve a display similar to what you hear. At a very fast attack setting the Analyser will show even shortest peaks down to one sample. But these are only audible when present in several bands and lasting for more than a millisecond. That's why 7 ms Rise Time results in a much better display behaviour. The release time should be set to three different values. A very fast display (1 s) is recommended for example when trying to see the bass which normally hides behind the bass drum. At 2 s the display will be more easier to read and is more friendly to your eyes. Some special applications (like averaging the sound impression or noise measurements) require values around 10 to 20 seconds.

Special Display Modes

The *Audio Analyser* offers two different envelope modes. Let's look at this example: You want to study the frequency distribution of a bass drum sample.

Method 1: Route the bass drum sample to the Analyser and store the display's contents by hitting *Freeze* in the right moment. Much more convenient is

Method 2: Use *Instant Hold* to activate the envelope mode. Play the bass drum sample one time. Thanks to the infinite release time the analyser's result won't vanish from the screen.

Method 3 is mainly used with music. At active *Instant Hold* and display mode *Peak Hold* the music is shown as usual, but the peak display above the bars won't be reset. When playing back a 3 minute Pop song an envelope of all frequencies and levels which occurred is shown.

Sound Measurements

The combination Equalizer/Analyser is used to correct the misalignment of speakers and room. A test signal (pink noise) is sent from the speakers into a special measurement microphone placed at the engineer's preferred listening position. At a release time of 20 seconds, Rise Time 200 ms and active *Point* mode a display of the overall frequency response is shown. The external Equalizer is now used to correct this curve to a straight line.

Special Applications

At the highest display range (50 dB) and lowest *Top Level* setting (-70 dB) the Analyser displays all signals down to -120 dBFS. This allows to examine the hum and noise present in the studio's environment. Disturbing frequencies can be classified and removed more exactly. In Left minus Right mode only the stereo signals of a complete mix are shown. This gives some interesting information on the mixing technique.

12. Controls and Connectors

Front panel



Pots for gain adjustment
0 to +20 dB

LC-Display

Keys and cursor control

Peak Level Meter Correlator

Rear panel

Word clock
In/Out



Mains connector

RS-232 interface

Digital outputs

Line inputs

Microphone inputs

13. Warranty

Before shipping each ADI-96 PRO is tested individual by RME in a complete test procedure. Using only the best hi-grade components allows us to offer full two years of warranty. The copy of the sales receipt or the Bill of Sale is your warranty legitimation.

The warranty does not cover damage due to abuse, incorrect installation or incorrect handling. RME's liability is limited to the repair or the replacement of the product, and does in no way include the liability for incidental or consequential damages resulting from use of ADI-96 PRO or its software.

14. Appendix

RME news, newest DSP-software and many information around our products can be found on our website:

<http://www.rme-audio.de>

Distributor in Germany:

Synthax Audio AG, Am Pfanderling 62, D-85778 Haimhausen, Tel.: (49) 08133 / 91810

CE

This device has been tested and found to comply with the limits of the European Council Directive on the approximation of the laws of the member states relating to electromagnetic compatibility (EMVG) according to EN 55022 class B and EN50082-1..

FCC Compliance Statement

Certified to comply with the limits for a Class B computing device according to subpart J or part 15 of FCC rules. See instructions if interference to radio reception is suspected.

FCC Warning

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation.

This device complies with part 15 of FCC rules. Operation is subject to the following two conditions:

1. This device may not cause harmful interference
2. This device must accept any interference received, including interference that may cause undesired operation.

However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna
- Increase the separation between the equipment and receiver
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected
- Consult the dealer or an experienced radio/TV technician for help.

In order for an installation of this product to maintain compliance with the limits for a Class B device, shielded cables must be used for the connection of any devices external to this product.