Digilyzer DL1

User Manual

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1. INTRODUCTION

Congratulations and thank you for buying NTI's Digilyzer DL1, a product specially suited for professional audio applications. The Digilyzer offers advanced analysis functions, expected only in much larger and more expensive systems. We are convinced you will enjoy using it!

NTI products are manufactured in compliance with the highest quality standards and marked with the CE sign.

CE Declaration of Conformity

We, the manufacturer

NTI AG
Im alten Riet 102
9494 Schaan
Liechtenstein, Europe

hereby declare that the product Digilyzer DL1, released in 2001, conforms to the following standards or other normative documents.

EMC-Directives:	89/336, 92/31, 93/68
Harmonized Standards:	EN 61326-1

This declaration becomes void in case of any changes on the product without written authorization by NTI.

Date: 01.11.2001

Signature: M. Becker

Position of signatory: Technical Director

International Warranty and Repair

International Warranty
NTI guarantees the Digilyzer DL1 and its components against defects in material or workmanship for a period of one year from the date of original purchase, and agrees to repair or to replace at its discretion any defective unit at no cost for either parts or labor during this period.

Restrictions
This warranty does not cover damages caused through accidents, misuse, lack of care, the attachment or installation of any components that were not provided with the product, loss of parts, connecting the instrument to a power supply, input signal voltage or connector type other than specified, or wrongly polarized batteries. In particular, no responsibility is granted for special, incidental or consequential damages.
This warranty becomes void if servicing or repairs of the product are performed by any party other than an authorized NTI service center or if the instrument has been opened in a manner other than specified in this manual.
No other warranty, written or verbal, is authorized by NTI. Except as otherwise stated in this warranty, NTI makes no representation or warranty of any kind, expressed or implied in law or in fact, including, without limitation, merchandising or fitting for any particular purpose and assumes no liability, either in tort, strict liability, contract or warranty for products.

Repair of your Digilyzer DL1

In case of malfunction, take - or ship prepaid - your NTI Digilyzer packed in the original box, to the authorized NTI representative in your country. For contact-details see the NTI web page.

Be sure to include a copy of your sales invoice as prove of purchase date. Transit damages are not covered by this warranty.

Warnings

In order to avoid any problems during the operation of the instrument, follow the rules listed below:

- **Use the instrument for the intended purpose only.**
- **Never connect the instrument to a high voltage output such as a power amplifier, mains power, etc.**
- **Do not disassemble the instrument.**
- **Never use the instrument in a damp environment.**
- **Remove the batteries as soon as they are flat or if the instrument is not intended to be used for a longer period of time.**

Test & Calibration Certificate

This is to certify the Digilyzer DL1 is fully tested to the manufacturer's specifications.

NTI recommends to calibrate this test instrument one (1) year after purchase. Thereafter the calibration- and adjustment interval is subsequently one (1) year.

2. OVERVIEW

The Digilyzer is a sophisticated tool used to analyze digital audio signals. It is designed for easy and quick maintenance and debugging of digital audio equipment and installations. Therefore, the Digilyzer observes the signals from many points of view simultaneously. An accurate overview of the actual signal condition is displayed on a large LCD and hidden errors are visualized (e.g. consistency check).

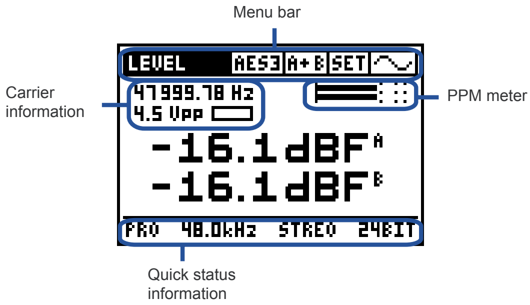

Fig 2-01, Overview display

By using the Digilyzer understanding and handling digital signals is simple. However, some basic knowledge about digital audio signals is essential. Please refer to our homepage for some easy literature about digital audio basics.

Interface types

The Digilyzer supports all common used interface types such as AES3, S/PDIF, TOS-LINK and ADAT. This range could be further extended by using external adapters or cheap equipment e.g. a TDIF to ADAT converter.

Functions

The Digilyzer features many useful measurement functions which are accessible through a menu bar.

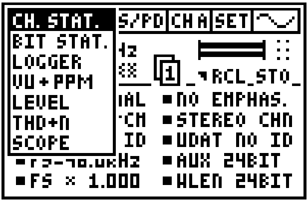

Fig 2-02, Measurement functions

Easy operation

The Digilyzer has an easy to use menu driven user interface. Changing of the settings takes place at the displayed values – there is no complex setup screen. The base element for operational functions is the cursor (inverted area) which can be navigated through the various screens by using the cursor keys. All selectable settings may be adjusted individually by pressing the enter key and selecting the requested value with the cursor keys. Confirm the setting by pressing the enter key again.

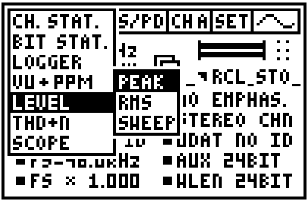

Fig 2-03, DL1 menu

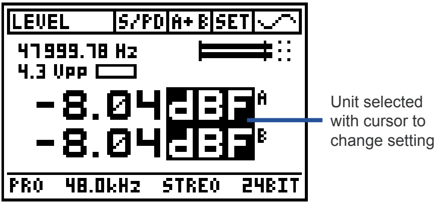

Fig 2-04, Changing settings

Monitoring

When it comes to digital audio, one of the major problems is, that humans do not have digital ears and thus can not listen to the incoming embedded audio signal. The Digilyzer offers a wide range of functionality to make digital audio audible.

What you hear is what you measure

- Functions with measurement of both channels (e.g. A+B), the output of the speaker is a mix of the two channel signals. Over the headphone output you may hear the signal in stereo (channel A on the left and channel B on the right side).

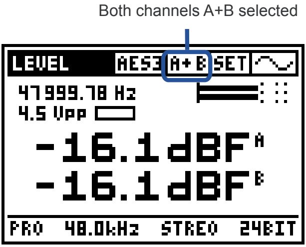

Fig 2-05, Channels A+B selected

- Some functions, e.g. SCOPE, are one channel functions only. You only hear the channel which is displayed on the screen.

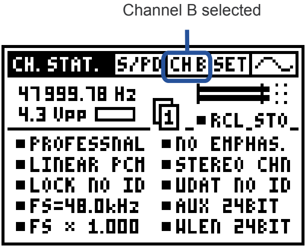

Fig 2-06, Channel B selected

Digital versus analog

Installations with digital and analog audio lines often cause headache and struggles. In reality some analog line maybe connected and the Digilyzer can not lock to the (non digital) signal. Therefore, the Digilyzer has the ability of analog monitoring. The Digilyzer may not lock to the input signal digitally, but it routes the signal directly to the speaker and headphone output, so you may hear the analog input signal acoustically. For visual clarification ANALOG MONITORING ON is flashing on the display.

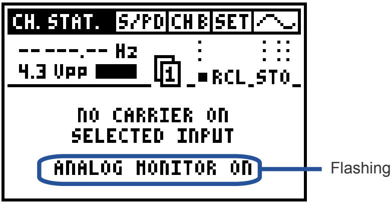

Fig 2-07, Analog monitoring

Debugging or listening?

The built-in speaker on the rear side gives the acoustical response of the measured signal – everywhere and any time – without the need of a headphone. For the users convenience a good headphone may be connected for listening in superior quality (up to 24 bit / 96 kHz). The headphone output may be used as D/A converter, e.g. some recording with a minidisc-player is requested but no analog output is available.

Oops – no signal!

Often no signal is found during troubleshooting at digital audio lines. Can this be? The Digilyzer offers a digital realized "AGC" (automatic gain control) to "zoom up" the digital signal. So even a change of the LSB (least significant bit) is fully audible. The tremendous dynamic range of **140 dB** allows to listen to the smallest disturbances. For example someone just has closed the fader or muted the signal, so just activate the AGC and listen; even dither noise will be audible.

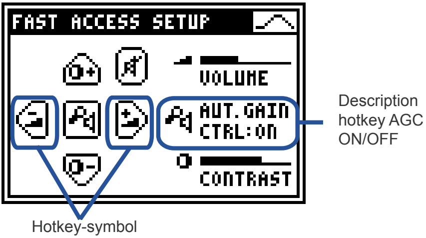

Fig 2-08, Automatic gain control

All the above monitoring functionalities can be controlled by using hotkeys, displayed in the FAST ACCESS SETUP menu.

Dual domain measurements

Level RMS and distortion (THD+N) measurements are fundamental to check A/D converters. The complete test system even checks mixed mode applications such as A/D converters or digital mixers is available in combination with the analog signal generator Minirator MR1.

Connectors

The Digilyzer includes the following connectors:

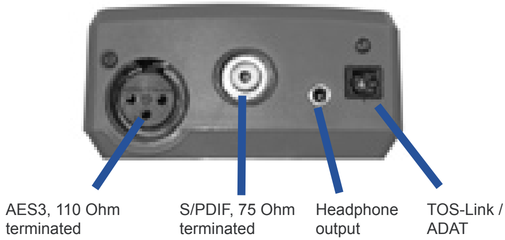

Fig 2-09, Inputs and outputs of DL1

NOTES

- **For AES3, 75 Ohm, signals (BNC connector) use the transformer AES3 75/100 Ohm, see accessories.**
- **Insert a jack into the headphone output to switch of the analog monitoring with the internal speaker (e.g. during a live act).**

Rubber Cushions

The original packaging of the Digilyzer includes a pair of rubber cushions. These may be adhered on the rear side of the device, so the output signal is made audible in good quality also whilst the Digilyzer lays, e.g. on a table, with the speaker on the bottom side.

Battery Replacement

Insert three pieces of 1.5 V alkaline batteries, type AA, LR6, AM3 into the Digilyzer battery compartment, as shown below.

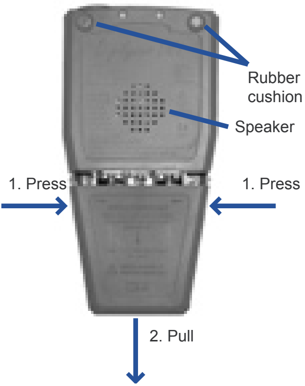

Fig 2-10, Open battery compartment

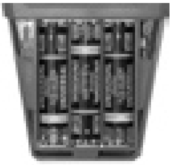

Fig 2-11, Inserted batteries

NOTE

- The use of rechargeable NiCd- or NiMH-batteries causes shorter battery lifetime than specified.
- Do not insert batteries of different types.
- Note the correct polarities of the inserted batteries.
- Remove the batteries as soon as they are flat.

3. FIRST STEPS

This chapter is a quick guide explaining how to make the first measurements with the Digilyzer. The example assumes an S/PDIF signal as input (e.g. a CD player with S/PDIF output playing a music CD).

1. Insert batteries

2. Connect the S/PDIF signal to the RCA input.

3. Reset the Digilyzer to default state by holding down the ESC key and simultaneously pressing the ON button for about two seconds.

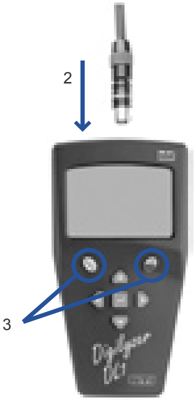

Fig 3-01, Digilyzer

4. Select S/PDIF as input format

- Move the cursor (inverted box) one step right to the format menu by using the cursor keys.

- Press the enter key to open the format menu.

- Select S/PD and press the enter key.

-> The channel status information is displayed and music is audible trough the built-in speaker.

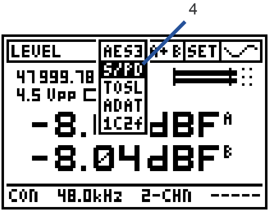

Fig 3-02, Format selection

5. Decrease the volume

   Press and hold the ESC- and left cursor key simultaneously. This causes the FAST ACCESS SETUP screen to be displayed and the volume to decrease.

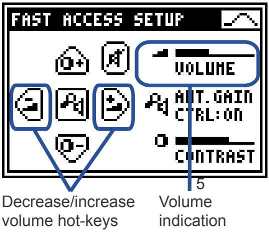

Fig 3-03, Volume setting

6. Measure the accuracy of the sampling rate

   The accuracy is measured in ppm, so the unit of the sampling frequency has to be changed as follows:

- Move the cursor to the unit “Hz”.
- Press the enter key to select.
- Press any cursor key to change from “Hz” to “ppm”.

-> The measured accuracy is immediately displayed.

- Press the enter key to confirm the new setting.

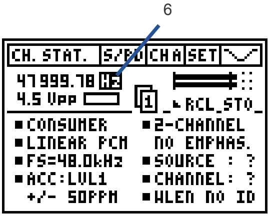

Fig 3-04, PPM setting

7. Activate the VU+PPM function

- Press the ESC key twice (cursor moves to the left top!).
- Press the enter key to open the measurement menu.
- Move the cursor down to VU+PPM.
- Press the enter key to confirm the new setting.

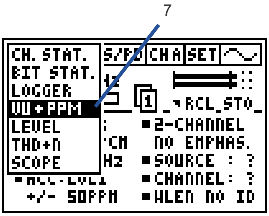

Fig 3-05, VU+PPM mode

8. Change auto power off time

- Move the cursor to SET on the menu line and press the enter key.
- Select AUTO POWER OFF with the cursor keys and press the enter key.
- Set the time to 60 MIN with the cursor keys.
- Confirm the setting with the enter key.
- Press ESC to leave the SETUP screen.

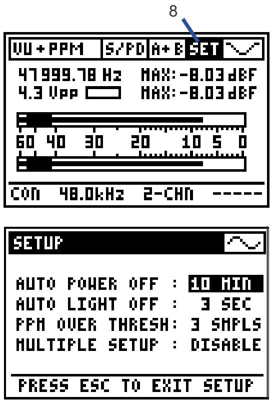

Fig 3-06, Auto power off setting

9. Backlight / power off

- Press the On/Off key shortly to energize the backlight.
- The backlight remains active according to the setting in the SETUP menu.
- Press the On/Off key for two seconds to switch the instrument off.

Congratulations, the first steps have been done with the Digilyzer to support the basic knowledge of the menu and device handling.

NOTE

- **Pressing the enter key changes a value directly or enters the selection mode (blinking cursor). The available settings may be selected with the cursor keys.**
- **In the selection mode you may press**
  - **ENTER to confirm the setting**
  - **ESC key to undo the setting.**

4. BASIC OPERATION

The operation of the Digilyzer is almost self-explanatory, despite the wide range of available measurement functions.

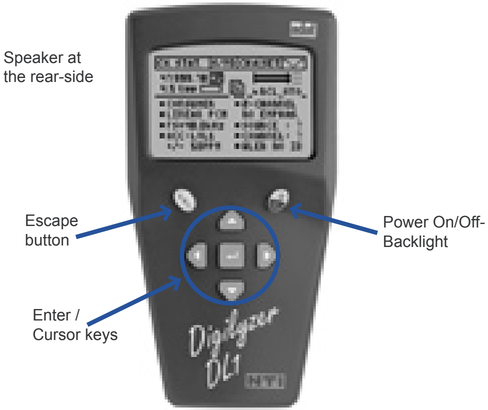

Fig 4-01, Control elements

The LCD is divided into the menu bar on top and the results displayed below, showing various information about the current status.

To quickly get the required information press the enter / cursor keys and the escape button to allow straightforward navigation through the available features. The cursor position is represented by an inverted display (white on black) of the field holding the cursor.

When the Digilyzer is on, it will utilize the same measurement function settings as switched off the last time.

Menu Bar

The menu bar allows the user to select the

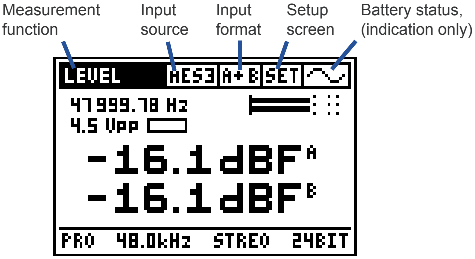

Fig 4-02, Menu bar

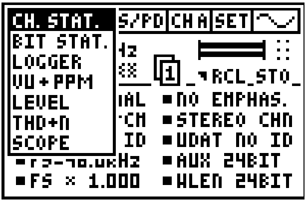

Fig 4-03, Measurement function menu

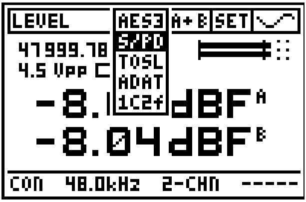

Fig 4-04, Input source menu

Any of the following formats may be selected

- AES3
- S/PD - abbreviation for S/PDIF
- TOSL - abbreviation for TOS-LINK
- ADAT - ADAT format via the TOS-Link input
- 1C2f - abbreviation of the single channel double sampling frequency mode, see details to this mode in the appendix.

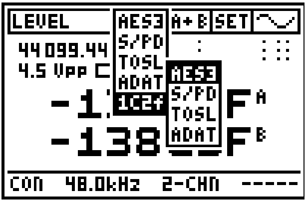

Fig 4-05, Format menu

Selection of Input Channel

Corresponding to the selected input source the available input channels may be selected. The individual measurement result of each channel is displayed. For a better understanding of the input channels:

- Channel A the left side of a headphone
- Channel B the right side of a headphone.

Indicator for Operation / Low Battery

A moving sine symbol indicates that the unit is running properly. Alternatively, this field shows a low battery indicator.

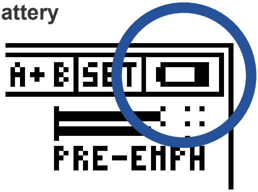

Fig 4-06, Low battery indicator

Setup Screen

The setup screen allows to customize basic settings of the Digilyzer.

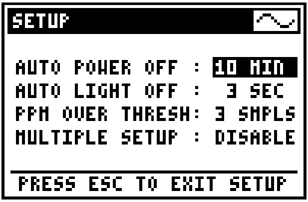

Fig 4-07, Setup screen

AUTO POWER OFF defines the time of the automatically switch off after the last key-press.

AUTO LIGHT OFF defines how long the backlight stays on after being activated.

PPM OVER THRESH defines the number of full scale values, causing a clipping indication at the ppm meter.

MULTIPLE SETUP allows four users to store their individual settings. To enable the multiple setup mode, set the corresponding entry to ENABLE. The next time the Digilyzer is switched on, the user will have to select the individual setup-ID (1, 2, 3 or 4) in the startup screen. All parameter settings in all measurement modes are now stored under this ID at the switch off and are recalled if this ID is selected at the next start up.

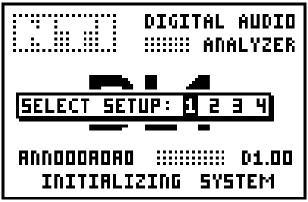

Fig 4-08, Multiple user startup screen

Fast Access Setup

Some key combinations allow fast access to the most frequently used settings. Pressing a hot key combination shows the FAST ACCESS SETUP screen and changes the value. If you don't remember the key combinations just press ESC for two seconds and the FAST ACCESS SETUP is displayed.

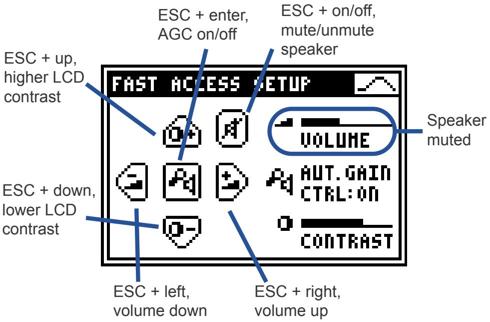

Fig 4-09, Fast access setup screen

The symbolized keypad on the left part of the screen indicates the hot keys function of the individual buttons. The actual adjustment is shown on the right side of the display.

VOLUME, setting of speaker volume. The volume control and the mute/unmute settings do not influence the analog monitoring. Analog audio signals are made audible with the speaker, but no measurement result is displayed. In this way analog audio signals are immediately indicated to the user. To analyze an analog signal the Minilyzer ML1 is recommended to be used.

AUT. GAIN CONTROL (AGC), all incoming signals are leveled to the same loudness, even if the signals are -60 dBF. The AGC amplifies a signal up to 140 dB, so even dithering noise on a silent line is audible!

CONTRAST, setting of the display contrast. For the detailed readout of fast changing display indications, e.g. SCOPE or VU+PPM, an increased LCD contrast is recommended to be set.

Display Basics

Debugging and analyzing digital audio signals need a few explanation to be best visualized at the same time. For example, a recorder does not recognize the incoming signal, so it is important to find out:

- Is there a digital or an analog signal on the line?
- Is there any audible information on the digital line?
- What is the format of the data (consumer / professional)?
- Does the signalized channel status correspond to the data?
- What's the level and the frequency of the carrier?
- … and much more

In many measurement functions the Digilyzer answers all these questions at the same time. For easy readout several screen elements are available on most functions.

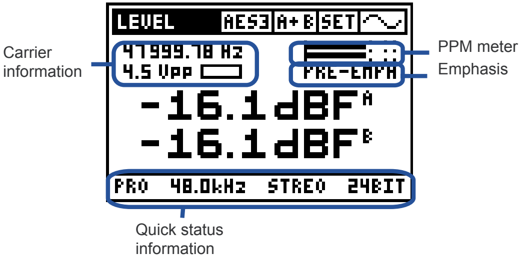

Fig 4-10, Screen elements

Carrier Information

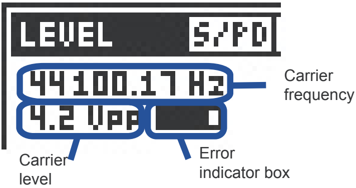

Fig 4-11, Carrier information

Carrier frequency displays the measured frequency in

- Hz
- ppm, deviation to the next standard frequency

Carrier level indicates the measured carrier level in Vpp. Levels higher than 5.0 Vpp are indicated as >5 Vpp.

Application hint:

The carrier level is a robust first approximation of signal quality. With decent short interconnection cables the carrier level

- of an AES3 signal is in the range of 2 to 7 Vpp
- of a S/PDIF signal is in the range of 200 to 700 mVpp

With impedance problems involved or with long cables the AES3 signal level may drop to levels below these values, so the reliability gets worse.

Error indicator box: The Digilyzer checks the digital signal, its protocol and recognizes a number of errors. These errors can cause many audible effects, which should ideally never occur. An error is immediately indicated by the Digilyzer by filling the error indicator box black. Afterwards the box empties within ten seconds.

Following errors are analyzed/visualized in the error indicator box:

- Lock / unlock
- Validity bit
- Confidence bit (the received data eye pattern opening is less than half of a bit period, indicating a poor link not meeting specs)
- Bi-phase mark coding error
- Parity error
- Carrier level below specification (for AES3 and S/PDIF input)

NOTE **The lock/unlock state is the only available error in the ADAT format.**

PPM Meter

Many measurement functions include a PPM meter, displaying the measured level peak in bargraph form. The scaling details are:

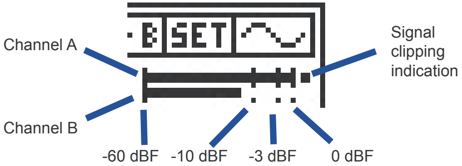

Fig 4-12, Bargraph markings

NOTE

- **For signal levels lower than -60 dBF the vertical line marked above with -60 dBF remains displayed and changes for muted signals to three dots.**
- **The number of full scale values causing clipping indication can be set in the SETUP menu (PPM OVER THRESH). By default it is set to three samples.**

Emphasis

The Digilyzer does not de-emphase any pre-emphased signals. In case the incoming signal is marked as pre-emphased in the channel status, the Digilyzer indicates this with PRE-EMPH displayed below the PPM meter (at all measurement functions with display space available).

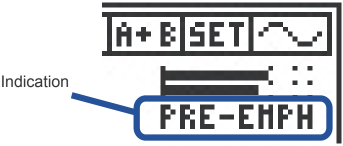

Fig 4-13, Emphasis

Quick Status Information

In the measurement functions BIT STAT., VU+PPM, LEVEL and THD+N the essentials of the channel status information are displayed continuously:

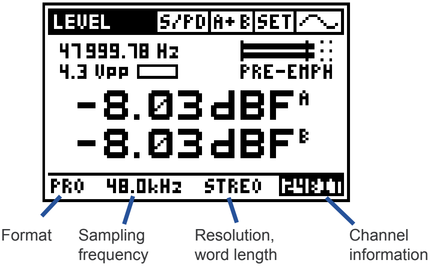

Fig 4-14, Quick status information

The consistency check is constantly running as a background task. It highlights any parameter showing inconsistency within the physical parameter. For example, as shown in the above figure, the resolution claims to be a 24 bit, but in reality it is lower.

5. MEASUREMENT FUNCTIONS

Channel Status

Digital audio signals (formats AES3 and S/PDIF) have additional information called channel status, encoded in the signal bit stream. The Digilyzer directly translates the contents of the status bits and displays the results on the channel status screen, thus enabling the user to directly read the meanings.
The proper interpretation of the status bits is carried out automatically by the Digilyzer. The first bit of the channel status indicates whether the status bits are configured in professional or consumer format. The professional format has various additional information encoded in the channel status, whereby in the consumer format, the copy protection is the major concern. ADAT signals do not carry any status information, so the display will indicate CHANNEL STATUS NOT AVAILABLE ON ADAT.

To display the complete status information three different pages are available:

- page 1, main status information
- page 2, additional status information
- page 3, status information in hex code

The page number is displayed at the top center of the measurement screen. Select the page number with the cursor and press enter to select the next page.

Professional Format

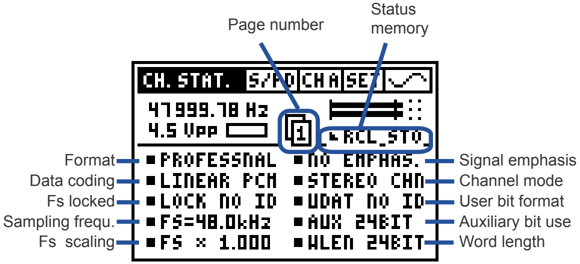

Fig 5-01 Channel status professional, page 1

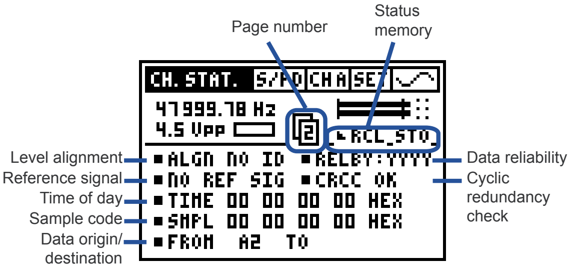

Fig 5-02 Channel status professional, page 2

Further details to the individual status indications are listed in the appendix.

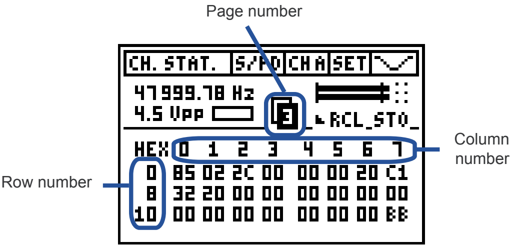

Fig 5-03, Channel status professional, page 3

On the third page the complete status information is displayed as hex code. The content of each status byte is shown as two digits in hex code. The status information contains 24 bytes. These are displayed within three rows and eight columns. The individual row- and column numbers shall be added together to read the information containing in the corresponding byte number.

e.g.	row number	+	column number	=	byte number
	0	+	4	=	4
	8	+	6	=	14
	10 (hex)	+	2	=	18

Application hint:

There are many undefined and reserved bit combinations in the channels status. The HEX view offers the possibility to further examine the reserved states - if necessary.

Consumer Format

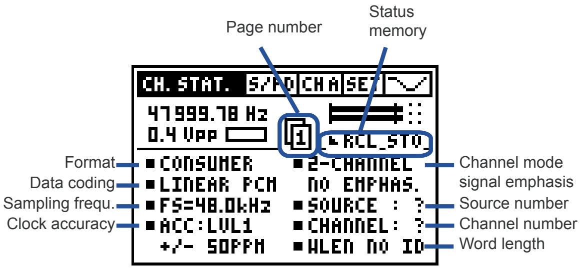

Fig 5-04, Channel status consumer, page 1

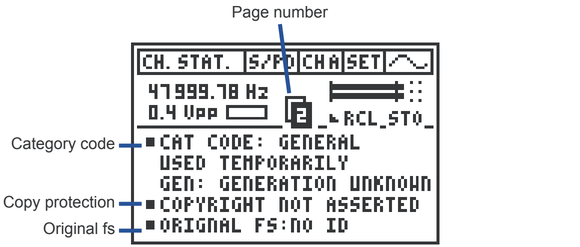

Fig 5-05, Channel status consumer, page 2

On page two are the fairly complex category tables, providing simple device statements, such as e.g. LASER OPTICAL PROD or MINI DISC SYSTEM, interpreted into words and letters for easy read out. The original sampling frequency field is used to indicate the fs of the signal prior sampling frequency conversion in a consumer playback format.

Page three displays the complete status information as hex code.

Consistency Check

The consistency check is constantly running as a background task. It compares the carrier information with the status information. E.g. the sampling frequency claims to be 44.1 kHz but in reality is 48.0 kHz. Such errors are immediately displayed via a flashing square around the individual status information.

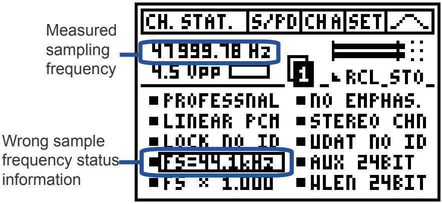

Fig 5-06, Channel status consistency check

The consistency check is carried out with the following parameters:

- Sample frequency
- Word length
- Clock accuracy
- 1C2f usage

Application hints:

- Wrong signalization of the sample frequency of digital audio devices may cause real trouble.
- Some units expected to be e.g. 24 bit devices, signalize 24 bit in the channel status but only send 22 relevant bits. The consistency check helps to see this problems quick and easy.

Channel Status Details

Detailed information about the interpretation of each individual bit and bytes may be found in the normative documents IEC 60958-3 and AES3. A summary is given in the appendix.

Channel Status Comparison

Both channels of an AES3 or S/PDIF signal have their individual channel status information. In 99% of all applications the content is identical. In case of any difference the small square indicators in front of each label would turn into triangles and constantly toggle.

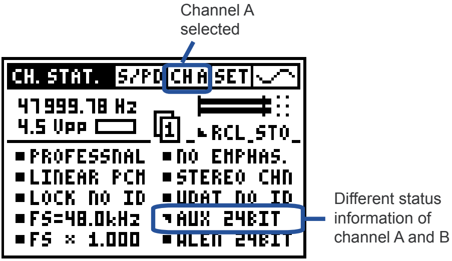

Fig 5-07, Channel status comparison

Channel Status Memory

The current channel status information may be stored and recalled.

Fig 5-08, Channel status memory field

Select STO with the cursor and press enter to store the actual status information. If any bit in the status is now altered compared to the currently stored status, the square flag on the left side of RCL will turn into the toggling triangle, indicating discrepancies. By selecting and holding RCL the memorized status information may be recalled for a quick check of the status difference. The status memory remains also valid after switching off the device.

Bit Statistic

The bit statistic function visualizes the state of all bits in the digital audio signal.

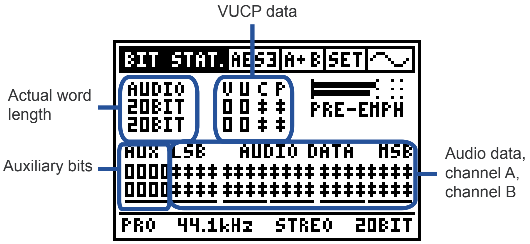

Fig 5-10, Bit statistic panel

The display allows you to see quickly which bits of the audio data are permanently low (0), high (1) or changing (indicated via the up/down arrow symbol).

Actual word length: The actually measured resolution is displayed.

VUCP data: The following bit information is displayed:

- V, validity bit indicating whether the digital audio bits may be converted to an analog audio signal; if the validity bit is permanently 0, the incoming data is valid.
- U, user bit containing any user bit information.
- C, status bit containing the channel status information, this bit is normally changing.
- P, parity bit, plausibility check of actual subframe, this bit is normally changing.

Auxiliary bits: These bits may be used for

- Audio data
- 2nd channel, e.g. talk back

Audio data: The two lines represent the 20 bit audio word for both channels.

- LSB on the left side, least significant bit
- MSB on the right side, most significant bit

The right bits always have to be active. In case some of the left bits are constantly 0, the resolution of the audio signal is obviously less than the maximum of 24 bits (including the aux. bits). The number of arrows from right to left may be counted to find the actual word length or binary resolution.

Application hints:

- Any digital input signal causes that some of the MSB's are active. The Digilyzer counts the number of active bits and displays the result as actual word length.
- Sometimes bits of the input signal are stucked on "0" or "1". In such a case a device in the signal chain is defective, e.g. a receiver or a transmitter.

Logger

The event logger records any irregularity of the digital signals. Besides the common expectations events such as change of sampling frequency, word length, consistency check results and many more are covered.

What is an event?

Events are changes or irregularities of the input signal. They are acquired for the two channels separately (wherever reasonable) and are split into the following categories:

- Carrier based events
- Frame based events
- Audio signal based events
- Channel status based events
- Consistency check based events

Basics of logging

Whilst a log is running, the Digilyzer acquires all events (details are listed in the Appendix). The Digilyzer bundles all events which are occurred within the recording interval. It generates a new “record” when a recording interval has finished. The recording interval can be adjusted at the start of a new log.

Basic setting of logger data

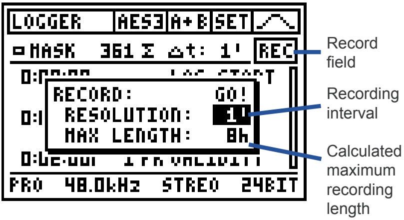

Fig 5-20, Recording setup window

- Select the REC field, press enter and the RECORD window is displayed.
- Select the recording interval of the recording. The max. recording length is defined by this resolution. The Digilyzer has the capability to store data of 500 recording intervals. A higher recording interval is causing a shorter maximum recording length.
- Select and confirm GO! to start the logger function. The recording setup window disappears and the REC field is flashing.
- The logging may be stopped by pressing the enter key at the REC field.

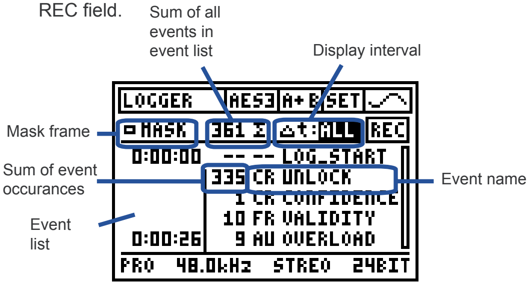

Fig 5-21, Overview records

Display interval: The events occurred during the user defined display interval are summarized and shown on the Digilyzer. The bundling of logger data helps to make the events more clear and easy to overview.

After the logging is completed the display interval may be changed.

- Zoom out the intervals, e.g. the error log has a recording interval of ten seconds and you want to know how many events have been found in the last hour, just zoom out to a display interval of one hour.
- Zoom in to get a more detailed information about the time of individual events occurred.
- Set to ALL (maximum zoom out) to get an overview of all records.

Displaying events

The display of the event logger is split into three areas:

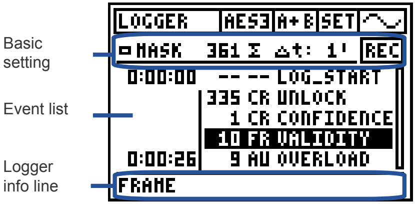

Fig 5-23, Detailed info of records

Event list: After the cursor is navigated to the event list, it can be scrolled with the up/down key. In this mode the data can be zoomed by using the left and right arrow keys. Press ESC to quit the event list.

Logger info line: While the cursor is in the event list, the individual detailed information to the selected event is displayed. Many errors may occur on channel A or B separately (e.g. AU OVERLOAD) and be indicated.

Masking events

For a better overview some events can be hidden from the logging display (e.g. the audio signal clips very often or the channel status changes permanently because of time code). Therefore, the Digilyzer offers you the possibility to select which events you want to see or hide. Select the field MASK to enter to the LOGGER DISPLAY MASK screen.

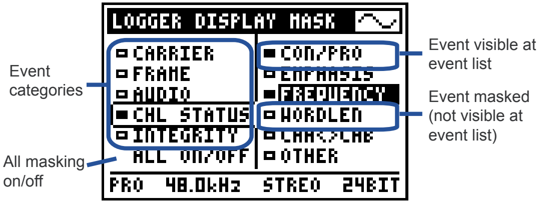

Fig 5-24, Logger display mask

The first column displays all event categories. The complete event category or single events of each category can be masked. The filled square indicates that the event is displayed at the event list.

NOTE **Masking does not effect the recording.**
All events are logged at anytime.

Overview of used event coding

See Appendix.

Application hint:

To read out one specific event only

- select in the LOGGER DISPLAY MASK the ALL ON/OFF field and press enter -> all squares will be empty
- select the specific event you are looking for, press enter and the specific square is filled up

VU+PPM

The Digilyzer offers a combined VU and PPM meter for two channels (stereo). This combination enables a quick and accurate overview on the momentary peak level and RMS value (power of the signal, which tends to be an indication for the volume).

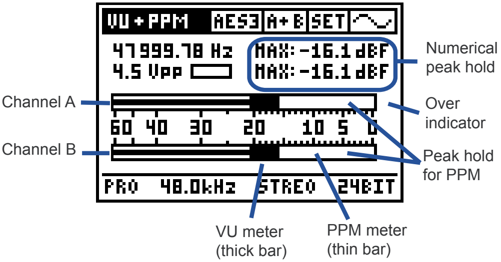

Fig 5-30, VU+PPM panel

Numerical peak hold indicates the all-time max. input peak level of each channel since the VU+PPM mode has been entered. It may be reset by placing the cursor on this value and pressing the return key.

Over indicator: Clipping indication; the number of full scale values causing this clipping indication may be adjusted in the SET menu (PPM OVER THRESH).

VU+PPM Indication: Digilyzer features

- VU, volume units indicated as the thick bar; displays the average volume level of the audio signal.
- PPM, peak program meter indicated as the center bar; displays the peak level of the audio signal.

Application hint:

Broadcast levels are limited to a maximum output peak level. This is in order to not to overload the transmission lines and to avoid unpleasant and audible distortions. During tuning through the radio stations it is clearly noticeable, that some channels are much louder than others; purposely to increase awareness!

This is accomplished by using compressors and other dynamic signal processing devices. The intention is to make the material as loud as possible without exceeding the maximum peak level. Simply feed the digital audio signal into the Digilyzer and it displays for both channels the peak- and VU-levels. The closer the VU level is to the PPM level, the higher is the compression of the audio material.

Level Measurement

The Level menu features three different measurement selections to choose from:

- Level peak
- Level RMS
- Level sweep

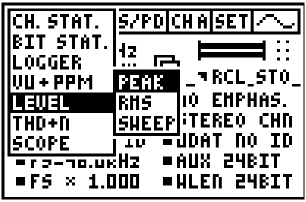

Fig 5-40, Level selection menu

Level peak measurements indicate – thought digitally – the maximum number (value) of the input signal.

Level RMS measures the power of the signal. The RMS functions of the Digilyzer such as Level RMS, sweep and THD+N are available as single channel functions only.

Level sweep, based on RMS measurement.

Application hint:

In digital audio the level peak measurement is basically mostly used, while in the analog area RMS values are important. Whenever you want to make measurements in the area of “frequency response”, RMS is usually the correct choice.

Level Peak

The Level peak function displays the actual peak value of the incoming digital audio signal. The digital peak level measurement provides information about the peak-to-peak signal level, compared to the full scale of the converter. The result is displayed simultaneously for both channels in numerical letters and in analog form, indicated in a bargraph.

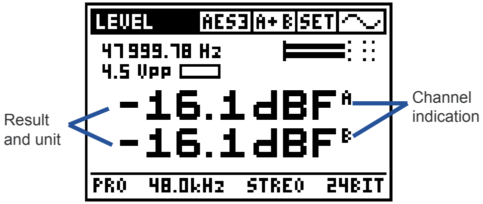

Fig 5-41, Level peak screen

Result and unit: The peak level can be displayed in three different units

- dBF (decibel full scale)
- % (percent of full scale)
- x1 (number, e.g. 0.1 of full scale)

The peak level units refer to the maximum possible level of the digital signal (100% or 0 dBF).

Application hint:

To measure the peak level during a period of time, use the VU+PPM function and monitor the numerical peak hold values.

Level RMS

The analog Level RMS function measures the RMS level of the digital input signal. Since the Digilyzer has no information about the reference voltage, such as the

- value to digitize the analog signal
- value to convert the digital signal into the analog domain

The RMS values are displayed as relative numbers compared to a sine wave signal with 0 dBF (peak value).

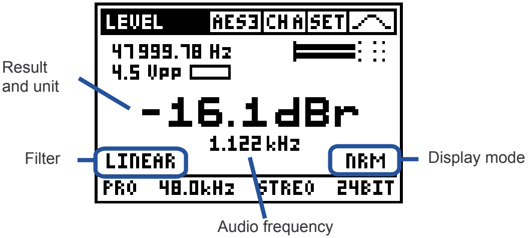

Fig 5-42, Level RMS

Result and unit: The RMS level of the individual selected channel can be displayed in three different units:

- dBr (decibel relative)
- % (percent of full scale)
- x1 (number)

In case the measured RMS value is smaller than -100 dBr, the Digilyzer displays "<-100 dBr".

Filter: The audio signal decoded from the digital audio stream can alternativelly be filtered with the following filters prior to the RMS or THD+N calculations:

- HP400, highpass 400 has a good rejection in the area of main frequencies – so hum problems may be easily examined and localized. The HP400 is also used to measure quantization noise.
- 22-22k, bandpass 22-22k is used to define the commonly used measurement bandwidth from 22Hz to 22kHz.

Display mode: Giving a better readability the display mode determines the rapidity of following up the input signal changes. The available modes are:

- SLOW 3 seconds averaging
- NRM 1 seconds averaging
- FAST no averaging

If averaging is active, measurements are smoothed in an exponential way (exponential time constant) before being displayed.

Level Sweep

The Digilyzer supports a RMS based frequency sweep. This function can be applied to measure the frequency response of devices.

During a frequency sweep, the Digilyzer records the Level RMS of every input signal, that has a stable frequency and level, provided that the frequency is higher than the one of the previous sample (otherwise the sample will be neglected).

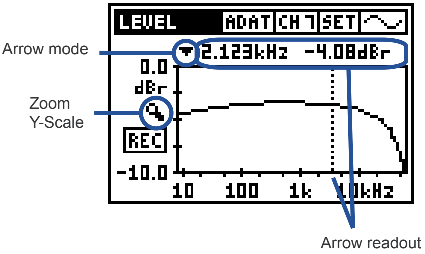

Fig 5-43, Frequency sweep graph

Within a graph every recorded sample is connected by a straight line approximately to the previous / next sample, thus building the displayed curve.

In practice the following steps are required / available for the execution of a frequency sweep.

- Arm the sweep recording process by moving the cursor to the REC field and pressing the enter key.
- The DL1 detects the start tone (315 Hz or 1 kHz) of an external sweep and the recording is automatically started. This status is indicated by the flashing REC field.
  Alternatively, the sweep recording may be started manually by pressing the enter key with the cursor on the ARM field. Consequently, the DL1 records every incoming signal with a higher frequency than the previous sample.

- The sweep recording will be stopped when as soon as an input signal with a lower frequency occurs or manually by the enter key pressed again (cursor on the flashing REC field).
- In order to analyze the sampled curve more detailed activate the arrow mode by placing the cursor to the corresponding symbol. Press the enter key and move the arrow to the sample(s) of interest by using the left / right keys.
- To zoom in/out the Y-axis move the cursor to the zoom mode field, press enter and use the left / right keys.
- To scroll through the Y-axis move the cursor to the zoom mode field, press enter and use the up / down keys.

The last recorded sweep curve will be stored internally, even after leaving the sweep mode or switching off the Digilyzer. As soon as the frequency sweep mode is re-entered, the curve will re-appear on the graph until a new frequency sweep is started.

NOTE **The auto power off is disabled during a frequency sweep recording.**

Test signal:

The sweep measurement can be carried out by using the sweep signal produced by the Minirator MR1, the "NTI Wavefile Generator" software or other external sweeps with the following pre-conditions:

- Signal type: Stepped sweep
- Step time: min. 1 second / step
- Frequency step: min. 1% increase
- Max. number of steps: 160
- Start frequency: 20 Hz

THD+N

The THD+N function (Total Harmonic Distortion+Noise) calculates the deviation of the input signal from an ideal sine wave. This measurement is most important to check and qualify analog/digital converters.

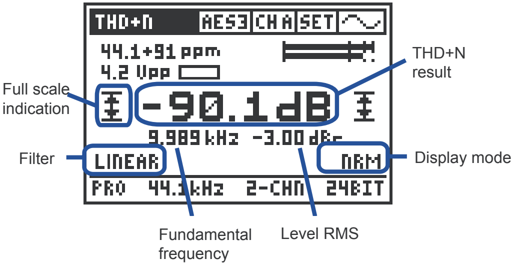

Fig 5-50, THD + N screen

The Digilyzer is able to calculate THD+N values down to –100dB (0.001%). For better THD+N values “< -100 dB“ (< 0.001%) are displayed.

THD+N results of the selected channel may be displayed in dB or %.

Full scale indication appears whenever one sample reaches full scale. This indication is independent of the clipping of the PPM meter.

Filter: The audio signal decoded from the digital audio stream can be filtered with the following filters prior to the RMS or THD+N calculations:

- HP400, highpass 400 has a good rejection in the area of main frequencies – so hum problems may easily be examined and localized. The HP400 is also used to measure quantization noise.
- 22-22k, bandpass 22-22k is used to define the commonly used measurement bandwidth from 22Hz to 22kHz.

Display mode: Giving a better readability the display mode determines the rapidity of following up the input signal changes. The available modes are:

- SLOW 3 sec. averaging
- NRM 1 sec. averaging
- FAST no averaging

If averaging is active measurements are smoothed in an exponential way (exponential time constant) before being displayed.

Application hints:

- Whenever one sample reaches full scale, slight clipping of the signal is possible so the THD+N value may degrade. Therefore, try to level the signal so the full scale indication does not appear.

- An analog to digital (A/D) converter may show the following errors at the signal conversion:
  - The imperfect linearity of the converter adds (hopefully little) new harmonics to the signal.
  - Every analog part generates noise which is added to the signal during conversion.
  - An A/D converter has only a finite resolution (e.g. 16 bit), so the converter must round each sample value, which results in an error called quantization noise.

  A perfect test signal fed into an ideal A/D converter causes a THD+N of the digitized signal of theoretically

  $-N * 6\ dB - 1.8\ dB$ N ... bit resolution of the converter

  E.g. a 16 bit converter has a theoretical THD+N of -97.8 dB. In practice good converters (even 24 bit) do not achieve better values than -110 dB. With such measurements the input signal is often the limiting point. To measure THD+N down to -100 dB a generated sine wave with a THD+N better than -100 dB is required. Such a sine wave is often generated only by expensive, special audio analyzing equipment.

Scope

The scope shows the waveform of the input signal. It measures the

- dominating fundamental frequency
- momentary peak level

and adjusts the X and Y-axis scaling automatically.

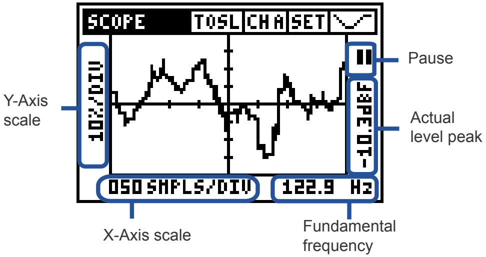

Fig 5-60, Scope screen

Y-Axis scale: Automatic scaling from 25%/div to 0.1ppm/div (allowing to see the LSB of a 24 bit signal).

Actual level peak: Since it is sometimes difficult to get a feeling for values of e.g. 0.6 ppm, the actual peak level of the data shown on the screen is displayed in dBF.

X-Axis scale: Automatic scaling from 1 to 500 samples per division.

Pause: The scope display may be paused by selecting this field with the cursor and pressing the enter key.

Fundamental frequency: The input signals that fundamental or most dominant frequency are displayed.

NOTE **The scaling of the SCOPE display cannot be changed manually.**

6. TROUBLESHOOTING

In case the Digilyzer is malfunctioning the software may be reset to factory set up as described below.

System Break Down

- Switch off the device.
- Reset the Digilyzer to the default status by pressing the ESC button and switching on the Digilyzer simultaneously.
- Release the ESC button.
- The below screenshot will appear on the display stating on the bottom line LOADING DEFAULT SETUP.
- Verify the correct operation.

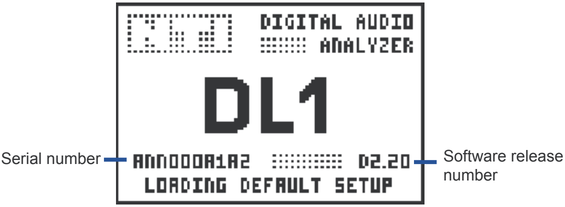

Fig 6-01, Start up screen loading default setup

In case you find system breakdowns happening several times or your device is malfunctioning, please note serial number and software release number and contact the local NTI representative in your country.

For contact details see the NTI web page:
www.nt-instruments.com

Monitoring

In two channel measurement functions (e.g. Level peak) the monitoring signal of channel A and B are mixed together (stereo). In case one of the channels is muted the stereo monitoring signal has a reduced level.

7. ACCESSORIES

MiniLINK

MiniLINK allows documentation and data acquisition of all DL1 functions in conjunction with the easy to use MiniLINK PC software.
MiniLINK is an upgradeable kit for all existing and new Minilyzers. It consists of a small plug-in USB interface board that can be easily installed without any tools. MiniLINK allows

- Storing measurement results and screenshots into the DL1 flash memory
- Logging on-line measurement results onto the PC

Fig 7-01, MiniLINK

Mains Power Adapter 7.5V (euro type)

The Digilyzer can be powered by batteries or by an external power. This power adapter is ideally suited for external power supply. Applicable for european connector types only.
Cable length = 2 meter.

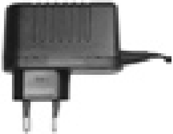

Fig 7-02, Mains power adapter

Transformer AES3 75/110

This digital audio transformer enables the measurement of the AES3 75 Ohm standard with the Digilyzer. AES3 75 Ohm standard is especially required for the transmission of digital signals over longer cables
(cable length >100 m).

Fig 7-03, Transformer AES3

Pouch

The soft pouch protects the Digilyzer against shocks, dust and water. With its convenient belt-clip you can keep it close to you even when you need both hands for other tasks.

Fig 7-04, Pouch

Minstruments System Case

Store your valuable Minstruments test system consisting of the Minirator MR1, the MiniSPL and the Minilyzer ML1 or Digilyzer adequately in the compact system case which gives you extra space for cables, connectors and other accessories you wish to bring along when you are ‘out in the field’ checking audio systems.

Fig 7-05, System case

8. APPENDIX

1C2f Format

The AES3 standard includes the following two options for 96 kHz sample rate operation:

- To double the frame rate from the previous 48 kHz to 96 kHz (not possible for older equipment). This is the normal operation of the Digilyzer.

- Using the two sub-frames (two channels) of a 48 kHz frame rate AES3 signal to carry consecutive samples of a mono signal resulting in a 96 kHz sample rate stream. The samples of one 96 kHz signal are packed “interleaved” into two 48 kHz signals. This allows older equipment, which transmitters and receivers are not rated for 96 kHz frame rate operation, to handle 96 kHz sample rate information.
  This mode is called “single channel double frequency mode” or “double wire mode” (two AES3 cables are needed for stereo). The Digilyzer also enables measurements in this mode. Just select the input format 1C2f and the input channel menu will indicate “A **i** B” (or eg. “1 **i** 2” for ADAT). This indicates that channel A and B are used in an interleaved manner (1C2f mode).

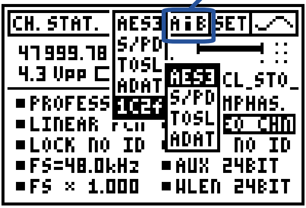

Fig 8-01, 1C2f selection

The Digilyzer is specified for samples rates up to 96 kHz, so for a 1C2f signal the resulting sample frequency shall not exceed this value.

NOTE

- **The second channel (e.g. B) is invalid in the 1C2f mode. This is indicated in the measurement functions like LEVEL PEAK as "—" or with a very small value.**

- **The 1C2f format is displayed as part of the channel status menu in the professional format. At many applications this information is not configured in the channel status details of the digital audio signal.**

- **The 1C2f mode is part of the consistency check. In case the channel status indicates 1C2f mode and 1C2f mode is not selected in the format menu or vice versa, the consistency error window is displayed.**

- **1C2f mode is defined for AES3 professional mode only. The Digilyzer supports the 1C2f mode also in the consumer and ADAT format. No monitoring is available for ADAT signals.**

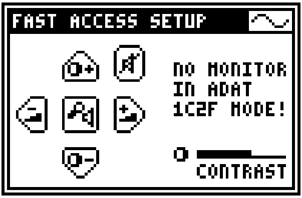

Fig 8-02, 1C2f with ADAT format

- **The DL1 allows you to switch to 1C2f mode even if the input sample rate is higher than 48 kHz. This may cause the DL1 to unlock to the input signal – or could exceed the processing power of the Digilyzer. This is indicated as "CARRIER FREQUENCY TO HIGH" in the quick status information on the bottom of the display.**

Logger Event Coding

The following chart lists all details to the Digilyzer Logger event coding. The remark column states:

- Any format not applicable (n.a.) for the individual events
- Maximum amount of events indicated per second (rec./s) or “sample“ (= each sample is counted)

DL1 event	description	remark
	Carrier based events	
CR UNLOCK **CR LOCK**	Digilyzer is not able to lock to the input.	10 rec./s
CR FS TO HIGH	Sample frequency in 1C2f mode is to high.	10 rec./s
CR CONFIDENCE	The received data eye pattern opening is less than half of a bit period (problems on the transmission line).	10 rec./s
CR BI-PHASE	Violation of the biphase-mark format of the carrier signal.	n.a. ADAT, 10 rec./s
CR LEVEL	A change of the carrier level greater than 100mV generates this event. Details about the carrier level (average, min. and max. carrier level) are also acquired.	n.a. ADAT &TOSlink, 1 rec./s
CR FREQUENCY	A change of the carrier frequency greater than 1 Hz generates this event. Details about the carrier frequency (min. and max. carrier frequency) are also acquired.	1 rec./s
	Frame based events	
FR VALIDITY	Validity bit set. This happens e.g. at a CD player with active error correction.	n.a. ADAT, sample

DL1 event	description	remark
FR PARITY	Parity error. The parity of the received signals is not correct (problems on the transmission line).	n.a. ADAT 10 rec./s
FR BLOCKCRCC	CRCC Error. The CRCC of the channel status information is incorrect (problems on the transmission line).	n.a. ADAT 10 rec./s
	Audio signal based events	
AU OVERLOAD	A clipping in the audio signal is detected. The clipping detector is configured in the Digilyzer setup screen. The setting for “PPM OVER THRESH” is stored within the log.	sample
AU MUTE	No audio signal but only zeros have been found.	10 rec./s
AU WORDLEN	The measured word length or the audio signal has changed (do not confuse with the word length signalized in the channels status).	10 rec./s
	Channel status based events	
CS CON/PRO	The professional / consumer indication bit in the channel status has changed.	n.a. ADAT, 10 rec./s
CS EMPHASIS	The emphasis bit of the channel status has changed.	n.a. ADAT, 10 rec./s
CS FREQUENCY	The signalized sample frequency in the channel status has changed.	n.a. ADAT, 10 rec./s

DL1 event	description	remark
CS WORDLEN	The signalized word length in the channel status has changed.	n.a. ADAT, 10 rec./s
CS CHA<>CHB	The status of channel A is not equal to the channel status of channel B.	n.a. ADAT, 10 rec./s
CS OTHER	Other bits than those mentioned in this table changed in the channel status.	n.a. ADAT, 10 rec./s
	Consistency check based events	
IC FREQUENCY	The sampling frequency indicated in the channel status is not equal to the measured sampling frequency.	n.a. ADAT, 1 rec./s
IC WORDLEN	The world length indicated in the channel status is not equal to the measured word length.	n.a. ADAT, 1 rec./s
IC FREQPPM	The measured accuracy of the sampling frequency is worse than the accuracy indicated in the channel status (consumer mode only).	n.a. ADAT, 1 rec./s
IC MODE1C2f	In the professional format the 1C2f mode should be signalized when used. This event occurs if the pro channel status signalizes a 1C2f mode but the Digilyzer is not set to 1C2f mode or vice versa.	n.a. ADAT, 1 rec./s

Professional Format Coding

Overview of codings and abbrevations used for the professional channel status display (MSB left).

Byte	Bit	Bit-info	Digilyzer Channel status	Digilyzer Quick view	Explanation
0	0	Use of channel status			
		0	CONSUMER	CON	Consumer format
		1	PROFESSNAL	PRO	Professional format
	1	Data coding			
		0	LINEAR PCM		Linear PCM samples
		1	NO LIN PCM		No lin. PCM samples
	2-4	Audio signal emphasis			
		000	EMPH NO ID		Emphasis no ID
		001	RES.EMPHAS		Reserved
		010	RES.EMPHAS		Reserved
		011	RES.EMPHAS		Reserved
		100	NO EMPHAS.		No emphasis
		101	RES.EMPHAS		Reserved
		110	50/15uS EM		50/15 ms emphasis
		111	CCITT EMPH		CCITT emphasis
	5	Locking of source sample frequency			
		0	LOCK NO ID		Locked (condition
		1	UNLOCKED		not indicated)
	6-7	Sampling frequency			
		00	see byte 4, bit 3-6		
		01	FS=48.0kHz	48.0kHz	
		10	FS=44.1kHz	44.1kHz	
		11	FS=32.0kHz	32.0kHz	
1	0-3	Channel mode			
		0000	CHNL NO ID	-----	Mode not indicated
		0001	TWO CHANNL	2-CHN	Two channel mode
		0010	SINGLE CHN	1-CHN	Single channel mode
		0011	PRM/SEC CH	PR/SE	Primary/sec. mode
		0100	STEREO CHN	STREO	Stereophonic mode
		0101	CH MOD RES	-----	Reserved
		0110	CH.MOD RES	-----	Reserved
		0111	1CH FS*2 M	FS*2	1C2f mode
		1000	1CH FS*2 L	FS*2L	1C2f, stereo left
		1001	1CH FS*2 R	FS*2R	1C2f, stereo right
		1010	CH.MOD RES	-----	Reserved

Byte	Bit	Bit-info	Digilyzer Channel status	Quick view	Explanation
1	0-3				
		1011	CH MOD RES	-----	Reserved
		1100	CH MOD RES	-----	Reserved
		1101	CH MOD RES	-----	Reserved
		1110	CH MOD RES	-----	Reserved
		1111	see byte 3		
	4-7	Userbits management			
		0000	UDAT NO ID		No user information
		0001	UDAT 192 B		192 bit block structure
		0010	UDAT AES18		AES18 standard
		0011	UDAT USRDF		User defined
		0100	UDAT 60958		Conforms to IEC
		others	UDAT RSRVD		Reserved
2	0-2	Use of auxiliary sample bits			
		000	AUX NO DEF		Use not defined
		001	AUX 24BIT		Use for main audio
		010	AUX TLKBCK		Use for talkback
		011	AUX USRDEF		User defined
		others	AUX RESRVD		Reserved
	3-5	Audio sample word length			
		000	WLEN NO ID	-----	applicable if byte 2,
		001	WLEN 23BIT	23BIT	bit 0-2 is “100“
		010	WLEN 22BIT	22BIT	
		011	WLEN 21BIT	21BIT	
		100	WLEN 20BIT	20BIT	
		101	WLEN 24BIT	24BIT	
		110	WLEN RSRVD	-----	Reserved
		111	WLEN RSRVD	-----	Reserved
		000	WLEN NO ID	-----	applicable in all
		001	WLEN 19BIT	19BIT	other cases
		010	WLEN 18BIT	18BIT	
		011	WLEN 17BIT	17BIT	
		100	WLEN 16BIT	16BIT	
		101	WLEN 20BIT	20BIT	
		110	WLEN RSRVD	-----	Reserved
		111	WLEN RSRVD	-----	Reserved
	6-7	Indication of alignment level			
		00	ALGN NO ID		Not indicated
		01	ALGN SMPTE		Acc. to SMPTE RP155
		10	ALGN EBU		According to EBU R68
		11	ALGN RSRVD		Reserved

Byte	Bit	Bit-info	Digilyzer Channel status	Digilyzer Quick view	Explanation
3	0-2	Channel identification			
		000	MCMD0 CH	MCMD0	Mode 0
		001	MCMD? CH	-----	Reserved
		010	MCMD2 CH	MCMD2	Mode 2
		011	MCMD? CH	-----	Reserved
		100	MCMD1 CH	MCMD1	Mode 1
		101	MCMD? CH	-----	Reserved
		110	MCMD3 CH	MCMD3	Mode 3
		111	MCUSR CH	MCUSR	User defined
4	0-1	Digital audio reference signal			
		00	NO REF SIG		No reference signal
		01	GRAD 1 REF		Grade 1 ref. signal
		10	GRAD 2 REF		Grade 2 ref. signal
		11	REFS RSRVD		Reserved
	3-6	Extended sampling frequency			
		0000	FS NO ID	FS_NOID	Not indicated
		0001	FS RESERVD	FS_RSVD	Reserved
		0010	FS RESERVD	FS_RSVD	Reserved
		0011	FS RESERVD	FS_RSVD	Reserved
		0100	fs=96kHz	96.0kHz	
		0101	fs=88.2kHz	88.2kHz	
		0110	FS RESERVD	FS_RSVD	Reserved
		0111	FS RESERVD	FS_RSVD	Reserved
		1000	fs=24kHz	24.0kHz	
		1001	fs=22050Hz	22050Hz	
		1010	FS RESERVD	FS_RSVD	Reserved
		1011	FS RESERVD	FS_RSVD	Reserved
		1100	fs=192kHz	192kHz	
		1001	fs=176400	176kHz	
		1110	FS RESERVD	FS_RSVD	Reserved
		1111	FS USERDEF	FS_USER	User defined
	7	Sampling frequency scaling flag			
		0	FS * 1.000		No scaling
		1	FS / 1.001		Scaling 1/1.001

Consumer Format Coding

Overview of codings and abbrevations used for the consumer channel status display (MSB left).

Byte	Bit	Bit-info	Digilyzer Channel status	Digilyzer Quick view	Explanation
0	0	Use of channel status			
		0	CONSUMER	CON	Consumer format
		1	PROFESSNAL	PRO	Professional format
	1	Data coding			
		0	LINEAR PCM		Linear PCM samples
		1	NO LIN PCM		No lin. PCM samples
	2	Copyright			
		0	COPYRIGHT ASSERTED		
		1	NO COPYRIGHT ASSERTED		
	3-5	Emphasis			
		000	2-CHANNEL NO EMPHAS.		Emphasis not indicated
		100	2-CHANNEL 50/15uS EM		50/15 ms emphasis
		others	RES FMTINF ----------		
1	0-7	Category code			
		Includes information about the equipment type, e.g.	MINI DISK SYSTEM, MD PLAYER / RECORDER, ...		
2	0-3	Source number			
		0000	SOURCE : ?		
		others	SOURCE : (number 1..15)		
	4-7	Channel number			
		0000	CHANNEL: ?		
		others	CHANNEL: (letter A..O)		
3	0-3	Sampling frequency			
		0010	FS=22050Hz	22050Hz	
		0000	FS=44.1kHz	44.1kHz	
		0001	FS=88.2kHz	88.2kHz	
		0011	FS=176400	176400	

Byte	Bit	Bit-info	Digilyzer Channel status	Quick view	Explanation
3	0-3	0110	FS=24.0kHz	24.0kHz	
		0100	FS=48.0kHz	48.0kHz	
		0101	FS=96.0kHz	96.0kHz	
		0111	FS=192kHz	192.0kHz	
		1100	FS=32.0kHz	32.0kHz	
		1000	FS NO ID	FS_NOID	
		others	FS RSRVD	FS_RESERVD	
3	4-5	Clock accuracy			
		00	ACC:LVL2 +-1000PPM		
		01	ACC:LVL3 VARIPITCH		
		10	ACC:LVL1 +/- 50PPM		
		11	ACC:RSRVD ---------		
4	1-3	Word length			
		000	WLEN NO ID	-----	applicable if byte 4, bit 0 is “1“
		001	WLEN 23BIT	23BIT	
		010	WLEN 22BIT	22BIT	
		011	WLEN 21BIT	21BIT	
		100	WLEN 20BIT	20BIT	
		101	WLEN 24BIT	24BIT	
		110	WLEN RSRVD	-----	
		111	WLEN RSRVD	-----	
		000	WLEN NO ID	-----	applicable if byte 4, bit 0 is “0“
		001	WLEN 19BIT	19BIT	
		010	WLEN 18BIT	18BIT	
		011	WLEN 17BIT	17BIT	
		100	WLEN 16BIT	16BIT	
		101	WLEN 20BIT	20BIT	
		110	WLEN RSRVD	-----	
		111	WLEN RSRVD	-----	
4	4-7	Original sampling frequency			
		1111	ORIGNAL FS=44.1kHz		
		1110	ORIGNAL FS=88.2kHz		
		1101	ORIGNAL FS=22050Hz		
		1100	ORIGNAL FS=176400		
		1011	ORIGNAL FS=48.0kHz		
		1010	ORIGNAL FS=96.0kHz		
		1001	ORIGNAL FS=24.0kHz		
		1000	ORIGNAL FS=192kHz		

Byte	Bit	Bit-info	Digilyzer Channel status	Digilyzer Quick view	Explanation
4	4-7	Original sampling frequency			
		0111	ORIGNAL FS RESVD_7		
		0110	ORIGNAL FS=8.0kHz		
		0101	ORIGNAL FS=11025Hz		
		0100	ORIGNAL FS=12.0kHz		
		0011	ORIGNAL FS=32.0kHz		
		0010	ORIGNAL FS RESVD_2		
		0001	ORIGNAL FS=16.0kHz		
		0000	ORIGNAL FS: NO ID		

9. TECHNICAL SPECIFICATION

Frame	Consumer / Professional, up to 24 bit, Sampling frequency f_s = 32 – 96 kHz, Also supports: Interleaved 96 kHz mode on all inputs (single channel double sampling frequency modes)
Measurements	
Signal	Level-FS, level-RMS, signal frequency, THD+N, event logger, frequency sweep, PPM, scope, overload detection, Full-Scale detection
Carrier	Sampling frequency (accuracy ±2.5 ppm), carrier level
Frame	Channel status according to AES 3 (ed. 2003) and IEC 60958-3 (ed. 2003), bit statistics
Event logger	
Recording resolution	selectable 1”, 10”, 1’, 10’, 1h
Recording duration	selectable 8’, 83’, 8h, 83h, 20days
Display Δt	1”, 10”, 1’, 10’, 1h, 10h, 24h, all
Recording intervals	max. 500
Single errors in one recording interval	10 - 36 000
Input connectors	AES 3 (110 Ohm) XLR, S/PDIF (RCA), TOS-Link, ADAT, AES 3id (75 Ohm) using optional adapter, phantom power resistant
Monitor	Built-in speaker, headphone connector
Display	Backlit graphic LCD

Power supply	- 3x AA size dry batteries (alkaline), typical lifetime 8 hours - External DC power, 7.5 VDC, 500 mA, stabilized, use NTI recommended parts only
Dimensions *(LxWxH)*	163 x 86 x 42 mm (6.4“ x 3.38“ x 1.63“)
Weight	300 g (10.5 oz) incl. batteries
Temperature	0° to +45° C (32° to 113° F)
Humidity	< 90 % R.H., non condensing

Quick Guide
Digilyzer


On / Off (2 sec)

Measurement Function Menu:

- CHANNEL STATUS
  - PAGE 1
  - PAGE 2
  - PAGE 3
- BIT STATISTIC
- LOGGER
- VU + PPM
- LEVEL
  - PEAK
  - REL
  - SWEEP
- THD + N
- SCOPE

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