d02

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digital dynamics processor

Manual

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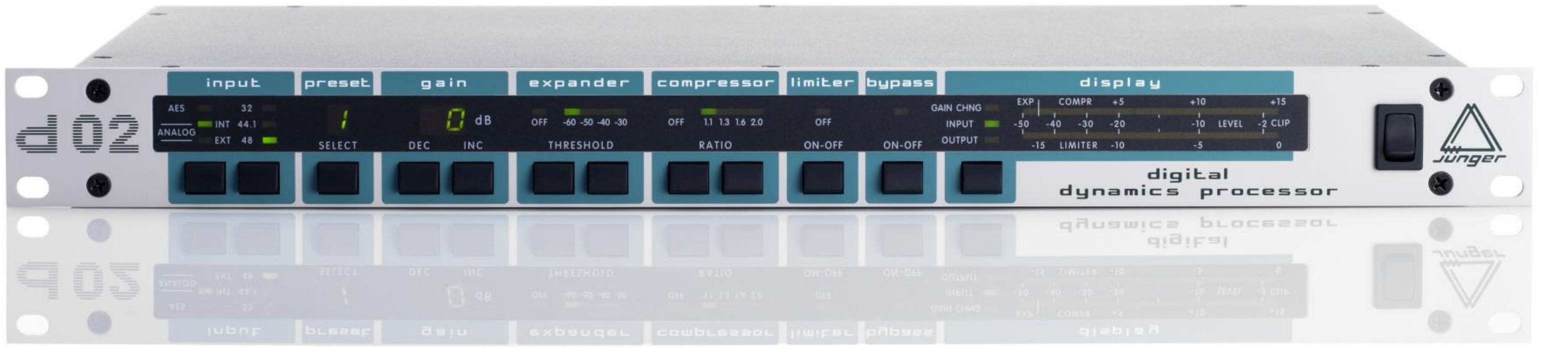

INTRODUCTION

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The **digital dynamics processor model d 02** is a professional studio device that processes the dynamic range of digital , as well as analog audio signals.

The unit comes with digital AES/EBU Interface and high resolution **24 Bit A/D Converters**, that allows dynamic range processing (compressor, limiter, expander) in the digital and analog domain.
The digital dynamics processor d 02 converts analog to digital audio signals without the risk of clipping and overload. With the combination of A/D-conversion (with headrom to avoid overload) and the following digital processing of **gain** and **limiter** it is possible to achieve the highest digital full scale signal without clipping.
The increase in programme density and loudness level are entirely free of the processing noises typical for dynamic range prossesors, such as pumping, breathing or signal discolouration.
The unit is easy to operate and requires only a limited selection of settings. All other parameters required for an inaudible processing of the dynamic range are automatically controlled by the programme signal and permanently optimized.

- fully **digital** processing device
  audio data word length: **24 bit**
- **compressor, expander, limiter**
- **4 presets** (universal, pop music, speech, live)
  for stereo or 2-channel-mode
  complex, signal dependent control algorithms
- linear **gain** - 6 dB ... +15 dB, in 1 dB steps
- digital **deemphasis filter**
- multicoloured **LED display**
  shows either input level, output level or gain change
  with peak hold and digital full scale display
- **digital audio interfaces**
  AES/EBU + S/PDIF + OPTICAL
- **analog input, analog output**
  24 bit over-sampling ADC, 24 bit oversampling DAC
  adjustable level, balanced
- redithering for 16 or 20 Bit digital output format

CONTENTS

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THE DESIGN OF THE DEVICE

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1.1. Basic Functions

The **d 02 digital dynamics processor** can be used to process both digital and analog audio signals. The device is primarily designed for use with stereo signals.
Digital input signals can be connected in the **AES/EBU** standard format, including SP/DIF and OPTICAL formats.
For the analog inputs high resolution **24 bit** A/D converters are used. The sample rate of the A/D-converter can be syncronised to internal crystal clock generators or to external word clock signals. Input and output can be selected independently. The output signals are available in parallel in all three digital formats so that, depending on the active input, a format conversion can also be achieved. In addition, an analog stereo signal output is available which operates with 24-bit D/A converters and enables a rapid acoustic monitoring.

The increase of signal density and loudness level of the digital audio signals can be achieved by the interaction of two dynamic range control processes. Firstly, by the **compression** achieved by increasing low and medium signal levels and secondly, by **linear amplification** combined with an inaudible **limitation** of individual remaining peak levels by the limiter.

The outstanding quality of dynamic range processing is based on the new **Multi-loop** dynamic range control principle developed by Jünger Audio.
The term **Multi-loop** means that there are several interactively combined control circuits as opposed to a control circuit with a spectrum split into several bands with different frequencies (multi-band).

1.2. The Jünger Audio Dynamics Processor Principle

A change in the dynamic range of an audio signal is a non-linear process. The gain of a dynamic range processor is not constant as it is with the gain of a linear amplifier. The gain varies in time depending on the input signal and depending on the specific control algorithm of the dynamics processor. These variations in the gain, which represent the real control process, should take place without any bothersome side effects such as pumping, signal distortion, sound colouration or noise modulation, which means they should be inaudible.

The main problem here is to react to fast changes in the audio signal (transients) without the control process being audible and disturbing. The ability of a dynamic range processor to react to rapid amplitude changes depends directly on its attack time. Long attack times do not cause modulation distortions, but lead to overshoots because the system is not fast enough to reduce the gain. A short attack time minimizes the amplitude and time of a possible overshoot, but a rapid gain change has audible side effects such as " clicks" caused by modulation products.

1. THE DESIGN OF THE DEVICE

traditional compressor and limiter designs

Traditional compressor and limiter designs only have one control circuit with corresponding attack and release times, which have to be adjusted manually by the user. An optimal setting of all parameters for dynamic range processing with as little disturbance as possible must be determined by listening and comparing.
A lot of experience and also a lot of time is necessary to get sufficient results. These parameters , once found, are only the right choice for a certain programme signal and must be changed for other signals.

multi-band structure

Dynamic range processors which split the audio frequency spectrum into several bands, i.e. which have a multi-band structure, have some advantages over traditional compressor designs. The dynamic control parameters in each band are independent of one another and can be set in such a way that a broad program range can be processed well. Disruptive side effects such as pumping and breathing can largely be avoided. The disadvantage of this system lies in the problem of rebuilding the output signal, which is the sum of all filters including those where dynamic changes have taken place as part of the control process.
The output signal is always coloured and deviates from the input signal in sound.

The dynamic range processor principle developed by Jünger Audio makes it possible to realise dynamics processors (compressor, limiter, expander) with very high audio quality, without signal discolouration, pumping or breathing, without distortion and modulation products - in short, with almost inaudible processing - and they are very easy to use.

multi-loop principle

The Jünger Audio dynamics processors work according to a Multi-loop principle, operating with an interaction between several frequency linear control circuits. The resulting attack and release times of this system are variable and adapted to the evolution of the input signal. This allows relatively long attack times during steady-state signal conditions but also very short attack times when there are impulsive input transients.

delay time

The Multi-loop structure also permits a short **time delay** between the control circuit and the gain changing element. The gain control circuit has time to preview the signal and become active before it reaches the output. This is particularly important for the limiter, which provides a precisely leveled output signal absolutely free of overshoots (clipping).

With a digital signal processor, a large number of parameters of the audio signal are evaluated and there is a permanent, automatic optimisation of the parameters of all control circuits.

Together with its attack and release times which determine the dynamic qualities, the performance of a dynamic range processor depends on the static compression characteristic.

The **d 02 digital dynamics processor** is a dynamic range processor which, contrary to its conventional counterparts, is effective for a wide dynamic range of input signals (50 dB).

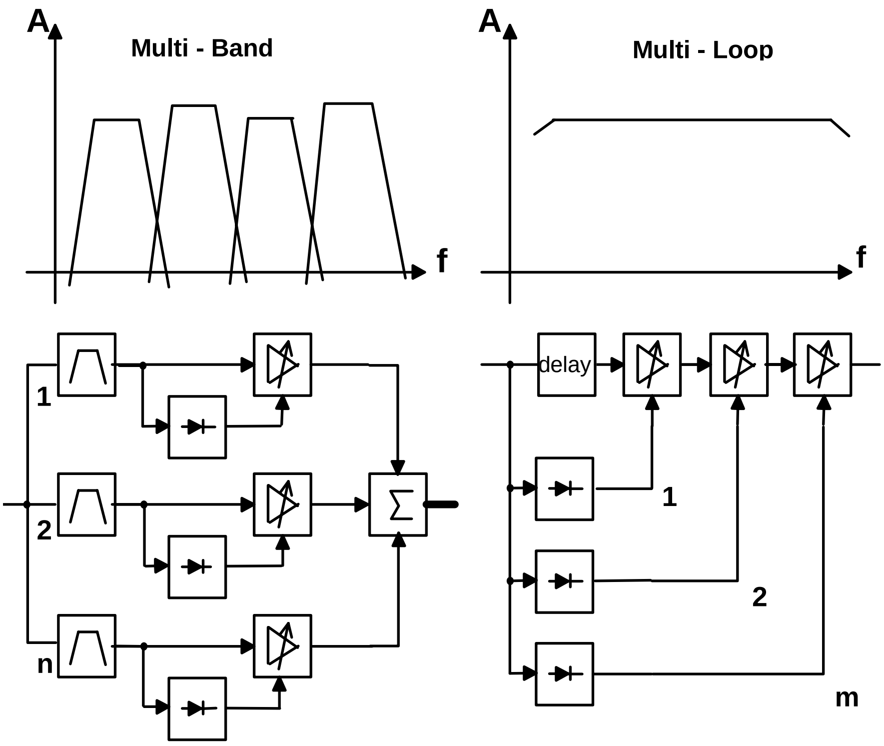


fig. 1:

basic principles of dynamic range processors

Figure 1 shows the basic principles of dynamic range processors.

compressor

The **compression** of the programme signal takes place evenly over the entire range and not only at the upper end above a certain threshold level. Dynamic structures of the input signal (e.g. musical dynamic evolutions) are converted proportionally so that even after compression the ratios are maintained, only slightly condensed, leaving on the whole a transparent, seemingly uncompressed s ound impression.

Compression (reduction of the dynamic range of the input signal to match the dynamic range of the storage or of the transmission system) is partly achieved by increasing the level of low level signals, the lowest of which might otherwise be below the noise floor of the audio system. The lower the input signal level the higher the additional gain applied to that input signal by the compression processing will be.

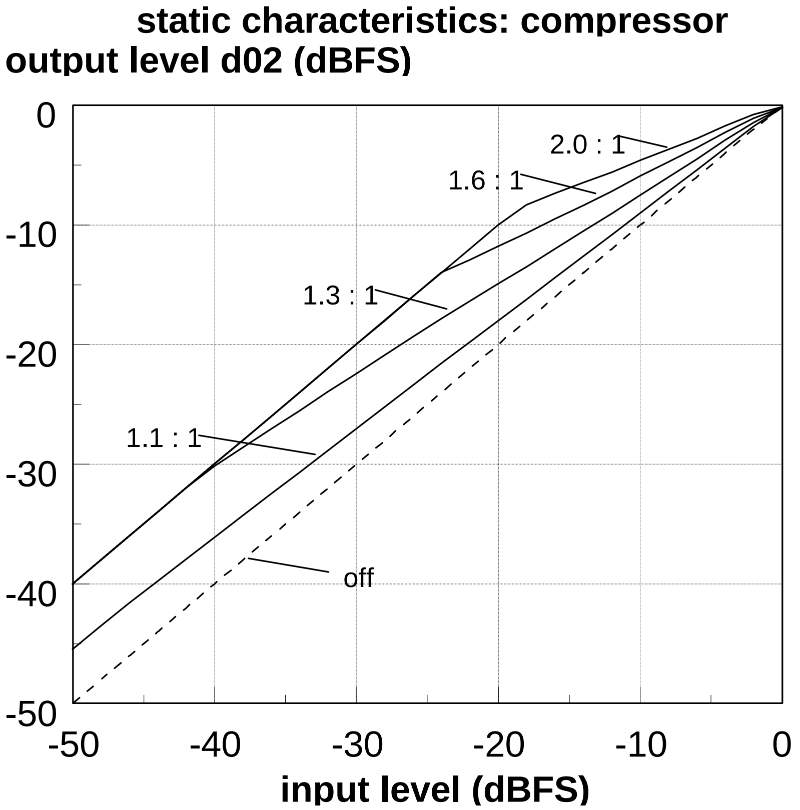
compression gain

Independent of the compression ratio , a **maximum gain of the compressor** can be set, so that there can be no inadmissible increase of background noises during signal pauses (e.g. live atmos, air-conditioning, hum and noise).

expander

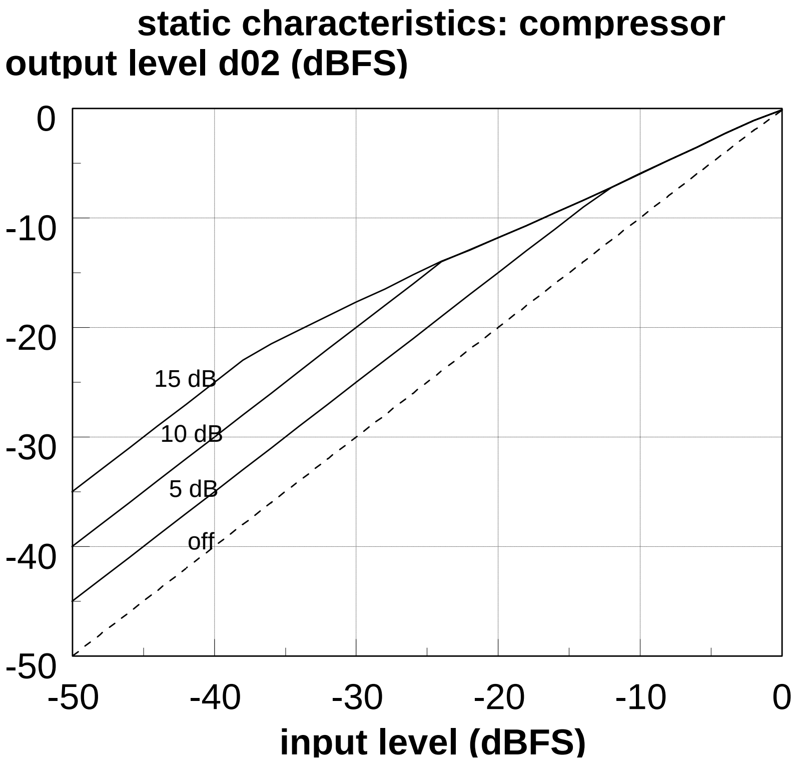
Below an adjustable threshold level an **expander** can be activated which can lower the amout of noise signals.

fig.. 2:
static characteristics: compressor



compression gain: max. 10 dB
parameter: ratio

fig. 3:
static characteristics: compressor



compression gain: max. 15 dB
parameter: compression gain
ratio: 1.6 : 1

1.3. A/D-Conversion with Digital Full Scale Level

The usable dynamic range for digital recording is determined at the top by the highest possible digital signal (full scale) and at the bottom by the lowest possible digital resolution. This range cannot be fully exploited when using a conventional analog-digital converter caused by the necessary headroom of 6 ... 10 dB to prevent over-level of the signal wich could otherwise occur.
This headroom of 6 .. 10 dB reduces the signal to noise ratio by the same amount even if a high quality A/D converter with 18 or 20 bit resolution is used.

It is therefore more important than noise-shaping or other dither techniques to use primarily the maximum of available digital dynamic range, because this improves most effectively the signal to noise ratio.

The d 02 digital dynamics processor offers a unique combination of a 24 bit A/D converter and a high quality digital limiter with which a digital signal free of overload and with maximum digital output level can be generated.

The A/D converter operates with normal headroom to avoid overload. Then in the digital domain the level of the signal is increased to the point where the limiter begins to control the level.
Any possible overload is corrected inaudible by the excellent audio quality of the digital limiter.

INSTALLATION

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2.1. Power Supply

The digital dynamics processor d02 is a device under the safety category *Schutzklasse 1* in keeping with the VDE 0804 standards and may only used with power supply installations built according to regulations.
Check the voltage details printed at the rear panel are the same as your local mains electricity supply.

2.2. Connections

All input and output connectors of the digital dynamics processor d02 are arranged in functional groups on the rear panel.

POWER INPUT
IEC mains input connector 100-240V, 50/60 Hz with integrated fuse

REMOTE
for optional serial remote interface RS-232 input and output
connector: 15pin SUB-D, male

DIGITAL INPUTS AND OUTPUTS

AES/EBU
input and output for AES/EBU standard format
input: XLR female panel jack
1- open, 2-3 signal, balanced, max. 5 Vpp
output: XLR male panel jack
1- open, 2-3 signal, balanced, max. 5 Vpp

S/PDIF
digital format for semi-professional use
When a signal is present at the AES input at the same time it has preference over SP/DIF
Input and output : RCA socket

OPTICAL
Optical interface for digital audio signals, (do not use input together with SP/DIF input)
Input and output : TOSLINK

EXT SYNC
Word clock input for external synchronisation
Input and output: BNC, (TTL-level)

ANALOG INPUT

Analog input to 24 bit A/D-converter
Input electronically balanced, XLR connector female
adjustable level (+12...+22 dBu for digital full scale)

ANALOG OUTPUT

Analog output from 24bit D/A-converter
Output electronically balanced, XLR connector male
adjustable level (+6...+22 dBu for digital full scale)

2.3. Switches for configuration of the unit

Following switches in the mode field at rear panel are used for configuration of the unit.

STATUS Setting of sended channel-status-bits on digital output by using of analogue input at any salmple rate.

Channel status bits are defined in the AES/EBU data stream. With the digital dynamics processor d02 it is possible to transmit this information without changing or to set these information defined.
(Sometimes it is helpful to change the channel status, f.i. if following units don't want to accept incoming signals.)

If using digital input of d02 unit is transparent for channel status information. There is no changing or modification of it possible. Channel status information at digital output is the same like original digital input signal.

PRO selection of professional mode.

CON selection of consumer mode.

DIG OUT Selection of dither mode for reduction of digital output word length.

16 BIT Dither for reduction to 16 bit word length

20 BIT Dither for reduction to 20 bit word length

24 BIT Signal without dither (unreduced 24 bit word length)

2.4. Setting the Digital Reference Level

The static characteristics of the processor d 02 are related to the digital reference level.
This internal digital reference level is the maximum output level for the limiter and the reference level for the static compressor characteristics. The rotation point for the compressor characteristics with zero gain is allways situated at the internal digital reference level.

In order to adjust the digital reference level it is necessary to change the operating mode of the unit as follows. Hold down the display button continuously for a few seconds and the unit will enter digital reference level adjustment mode. Pressing the INC or DEC buttons on can change the digital reference level in the range of 0 dBFS till -15 dBFS.

It is possible to store two different digital reference levels, one for use when the analog signal input is selected and another different setting for use when the digital input signal is selected. When changing the input selection between analog and digital the required reference level setting is automatically selected. So it is very easy to optimize levelling and headroom of the model d02 for different applications in analogue or digital mode.

For a digital mastering and transmission the output level should be the maximum, i.e. the digital reference level should be 0dBFS.

When working with analog inputs it is very important not to overload the A/D convertor (ADC), in order to ensure that the ADC always provides accurate linear conversion of the analogue input signal to the digital audio signal which is used for internal processing.
The analog input gain of the d02 should be set so that the maximum possible studio output level which will occur in practice must not overload the A/D converter.
When using the analogue output the analogue output gain following the digital to analogue converter must also be adjusted so that the internal digital reference level (maximum digital level which can be output by the digital limiter) corresponds to the maximum analogue level desired for the recorder or transmission line. Input a continuous signal such as a tone which is large enough for the limiter to start to operate and for the maximum output level to be output. The level on the d02 output level meter should correspond to the internal digital reference level which was set. Then adjust the analogue output gain to get the desired maximum analogue output level.

The calibration of the reference level should meet the maximum level of the transmission line or the transmitter. The internal reference level (limiter maximum output level) is always the absolute maximum level which the d02 will output.

CONTROL AND DISPLAY ELEMENTS

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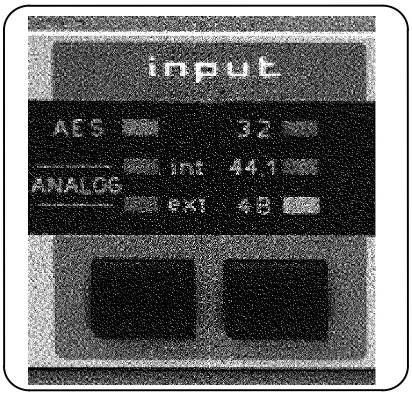
All functions of the **digital dynamics processor d 02** are activated by buttons. The front panel shows easily recognizable function groups.



input

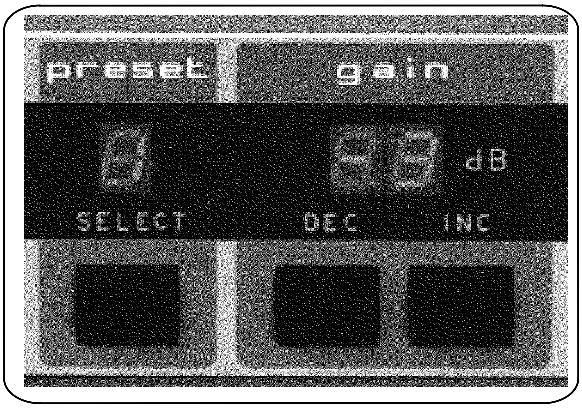
By pressing the left button in the input section the required input signal can be selected. Each time the button is pressed the input selection is changed and one of the three LED's above the button lights to show the newly selected input.

When the AES LED is lit the unit processes the AES/EBU format digital audio signal applied to its AES/EBU input connector.

When ANALOG INT LED is lit the unit processes the analogue input audio signal aplied to its analogue input connectors, and the sampling frequency at which the A/D-converter operates is generated internally.
When ANALOG EXT LED is lit the unit uses the same analogue input audio signal as when ANALOG INT is selected, but now the sampling frequency at which the A/D converter operates is determined by the external word clock or AES/EBU input signal which is fed into the unit.

To the right of the input indicator are three LEDs which shows the **sample rate** of the selected input. If a given external digital signal (input signal or wordclock) has the correct sample rate, the device automatically synchronizes to that frequency and a **yellow** light appears on the LED. All LEDs will blink **red** if the input signal is lacking or the sample rate is outside the admissible tolerance range.

With internal synchronisation (ANALOG intern) the sample rate display is **green** and the frequency can be changed with the button below.



preset

Press the PRESET button to select the one of the four operating programs of the unit which best corresponds to the kind of audio programme material which is being processed. Each operating program has optimum values of dynamic control characteristics (such as attack and release times etc.) for a different type of programme material.

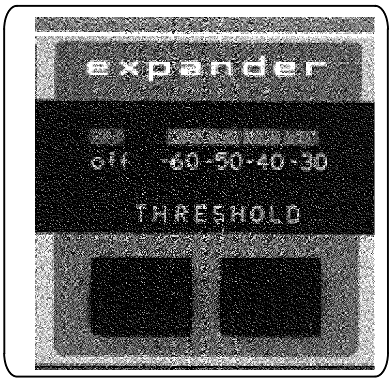
in stereo mode (loop function)	in 2-channel mode (loop function)
1 - universal	5 - universal
2 - popl music	6 - pop music
3 - speech	7 - speech
4 - live	8 - live

To change preset group hold down the display button continuously for a few seconds and the unit will enter the stereo/2-channel setting and the internal digital reference level setting mode. The PRESET and the GAIN display flashes and and the GAIN display shows the digital reference level.

The STEREO/2-CHANNEL mode can now be changed pressing the SELECT button. With every tip the unit toggles between the selected program in stereo or 2-channel mode. If you leave this setting function you can select your working program like described above.

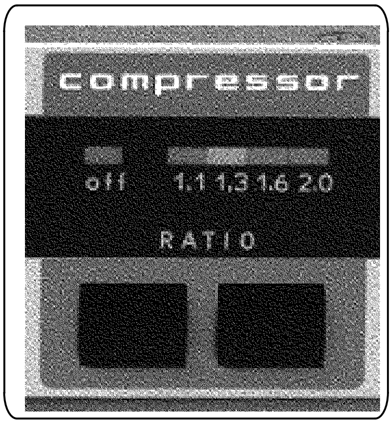
gain

The **INC**rement and **DEC**rement buttons allow a linear amplification of the digital input signal. The selection of gain levels takes place in steps of 1 dB and has a range from -6 dB ... +15 dB. Each time the button is pushed there is a change of 1 dB. Holding down the INC or DEC button continously leads to a continuous change in gain until the respective end value is obtained. When the gain level reaches **0 dB** there is a short pause to avoid negative gain (attenuation) being accidentally activated.



expander

The expander **THRESHOLD** can be changed upward or downward with two buttons and is visible on the LEDs above them. Four expander thresholds (-60 dB, -50 dB, -40 dB, -30 dB) can be selected. The threshold level is related to the choosed digital reference level.
In the OFF position the expander function is switched off.
The activity of the expander is indicated with a **red** LED in the **display gain reduction**.

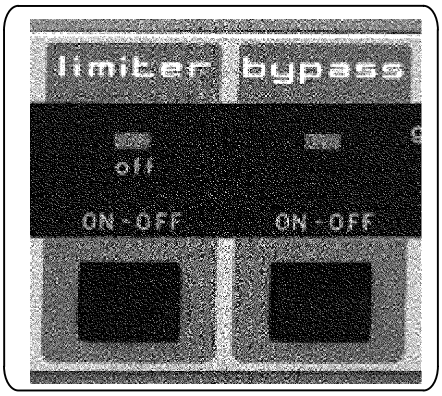


compressor

The compression ratio is adjusted by pressing the RATIO button and the currently selected ratio is shown by the lighting of the appropriate LED above the RATIO button.
One of four different ratios can be selected (1.1 : 1, 1.3: 1, 1.6 : 1, or 2 : 1). There is also a compressor off position where the compressor function is turned off. In this case none of the ratio LEDs will be lit.
Compression is partly achieved by increasing the level of low level signals, (the lowest of which might otherwise be below the noise floor of the FM transmission system). The lower the input signal level the higher the additional gain applied to that input signal by the compression processing will be. The **maximum amount of gain** applied to a low level signal can be adjusted independently of the compression ratio. Press both the RATIO buttons at the same time until normal gain display will be switched off.

maximum compression gain

A red LED will light in the compressor gain display which indicates the maximum value. This value can be changed with the keys **INC** and **DEC** in the range of 2 dB ... 15 dB.

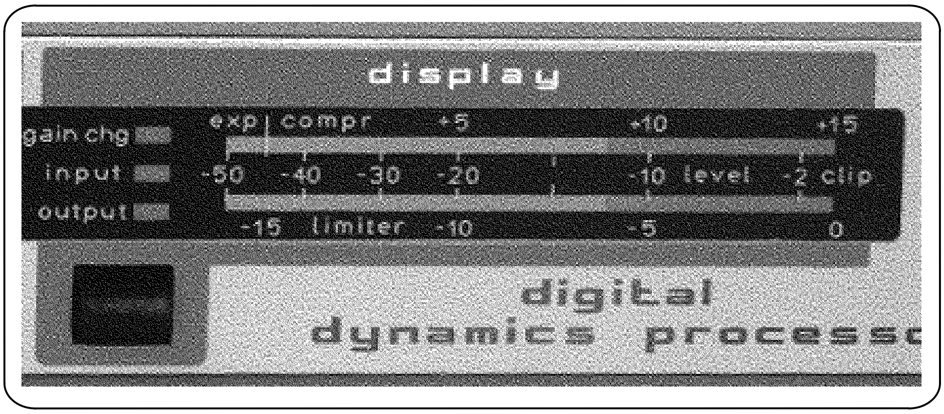


limiter

The limiter limits the maximum output signal level of the d02 precisely to the set **digital reference level**. (see also 2.4., and, for details of setting the digital reference level, see under "display"). The limiter should be always active to ensure that output level of the d02 never exceeds the preset digital reference level.
The LED shows a **red** warning signal when the limiter is turned **off**.
The limiter works with a look ahead time (signal delay) of approx. 2 ms.This delay time is present even when the limiter is turned off.
Two different reference levels can be set, one reference level for use when using a digital input signal, and another for use when using analogue input.

bypass

In the bypass mode (corresponding LED lits **red**) the digital signal is passing unprocessed through the DSP to the output. The signal delay time of approx. 2 ms is also effective in bypass mode.
The bypass function is not a relay bypass and is therefore not effective when the device is turned off from mains power.



display

The two channel LED display has three display modes (input level, output level and gain change). Press the button in the display section to change the display mode. The selected display mode is indicated by the lighting of the appropriate LED above the display button and to the left of the display meters. For better visibility each display mode has its own LED colour and level meter colour.
Green shows the **input level** and **yellow** the **output level**. The scale located between the two bars indicates the levels. The display which ranges from -50 ... 0 dBFS (dB Full Scale) refers to the digital reference level, with a resolution of 2 dB in the upper section. This does not allow a precise adjustment, but it does give an indication of the existence and the level of digital input and output signals.
A **peak hold** function is available for input and output which makes improved registration of a momentary peak level possible.
If excessive level at the input occurs when the input level display is selected (if digital audio samples at the maximum permissable positive or negative sample value occur at the digital input) then the red clip-LED at the extreme right-hand end of the level meter lights up and indicates overloads which are already present in the input signal.
When viewing the OUTPUT level the clipping LED does not light since the limiter is ON and ensures that the maximum output signal level can not exceed the preset reference level.
The level meter display is a digital meter without integration time, and records every successive digital sample value.

The third display mode, gain change, shows the current control levels of the limiter and compressor in dB.
The compressor works to reduce overall dynamic range by insertion of additional gain for lower level signals (ie no gain reduction). The scale above the upper meter bar shows the additonal gain inserted by the compressor. Lighting of LED's in the meter starts on the left and moves torwards the right as more additional gain is applied.
The limiter works to reduce the level of high level input signals so that they do not exceed the preset reference maximum level. The scale below the lower meter bar shows the level reduction by the limiter. Lighting of LED's in the meter starts on the right and moves torwards the left as the amount of level reduction (limiting) required increases.
A red LED is visible in the compressor gain display , which indicates the maximum permissable value of compressor gain. This value can be changed in the range +2dB to +15dB (see section on operation of the compressor on page 10 for details of how to change the maximum permissable compressor gain).

Setup selections using display key

The DISPLAY button has a second function in addtion to changing the display mode. It is used for setting the internal digital reference level, which is the maximum output level which the limiter will allow to be output by the unit. Hold down the display button continuously for a few seconds and the unit will enter internal digital reference level setting mode. The GAIN display flashes and shows the digital reference level.

The maximum output level permissable for the unit (internal digital reference level) can now be changed in 1dB steps within the range -15dBFs to 0dBFs by pressing the INC and DEC buttons. The reference level to be used when using the analog input and the reference level to be used when using the digital input can be set independently.

FUNCTIONAL DESCRIPTION

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Power-on Setting

After switching the power on, the **digital dynamics processor d02** automatically chooses the settings used before the power was turned off.
All parameters used, e.g. input, preset, gain, compressor, expander and display, are stored and re-applied. The only exception is the limiter which, as a safety function, is always activated when the power is switched on.

Digital input signals

The device is capable of processing digital audio signals as well as analog audio signals. The unit accepts an AES/EBU format digital audio signal. In the case of a digital audio input signal being processed the internal sampling frequency of the unit is automatically synchronised to that of the digital input signal. The sampling frequency may be any frequency in the range 30KHz to 50KHz. The d02 directly measures the actual sampling frequency of the AES/EBU input signal with a frequency counter. It does not rely on the indicated sampling frequency of the AES/EBU input signal, which is contained in the signals "channel status" data, being correct.

Digital input signals - sample frequency

If the measured input signal sampling frequency is one of the standard frequencies (32kHz, 44.1kHz or 48kHz) then a corrresponding LED will light yellow in the input section on the units front panel. Continuous lighting yellow of an LED also indicates that the digital input signal is a valid AES/EBU digital audio signal which the d02 can synchronise to properly.

Digital input signals - AES/EBU

If AES digital input is selected but the d02 can not synchronise properly to a supplied AES/EBU input signal (for example because there is no valid input signal or because the input signal has a sampling frequency outside the admissable tolerance range) then all three "sampling frequency" LED's in the input section of the d02 front panel will flash red. Digital audio input signals in the standard AES/EBU format pass from the AES/EBU input connector through a transformer (as specified by the AES/EBU standard) to the AES/EBU interface circuitry. The AES/EBU input circuitry derives the d02's internal sampling frequency from the AES/EBU input signal and seperates the audio data in the AES/EBU bit stream from additional control bits, such as channel status data bits (C-bit) and user bits (U-bit). The audio sample data is converted from AES/EBU format into the d02's internal digital format for processing. Data in AES/EBU control bits (C-Bit, U-Bit) will be passed from the AES/EBU digital input to the AES/EBU digital output unchanged.

Digital input signals - S/PDIF

The processing of digital audio data in the consumer format **S/PDIF** is also possible. If signals are present at both the AES/EBU and the S/PDIF inputs at the same time, the AES signal automatically has priority.

A/D-converter

Analog audio input signals can be fed into the unit via the analog input XLR connectors and first pass through an electronically balanced analog input amplifier, then to an Analog to Digital converter (ADC). The gain of the electronically balanced analogue input amplifier can be adjusted, using potentiometers on the rear panel. The maximum analog input level , which will correspond to a digital full scale (0dBFs) digital output signal from the internal ADC, can be set to any value in the range +12dBu to +22dBu.
The standard factory setting of the unit when supplied is that a programme level of +6dBu at the analog input corresponds to -9dBFs (9dB below maximum possible digitally represented level which can be output by the A/D convertor). Therefore the maximum permissable analog input level without clipping when the unit is supplied is +15dBu.
Both analog inputs are converted into digital audio signals, which can then be processed by the internal digital dynamics processing. Conversion is done by a high performance 24 bit oversampling A/D converter which is manufactured by CRYSTAL Semiconductor. The analogue to digital converter has a dynamic range of 114dB and is very linear in terms of both frequency and phase response. Provided that the maximum permissable analog input level (which will correspond to 0dBFs (full scale) internal digital input level shown on the units level meters) is not exceeded the A/D conversion process should have no signficant influence on the sound quality. The audio sample data output from the A/D converter is converted into the d02's internal digital format for processing.

Sample rate

When the input is set to 'ANALOG intern' the sampling frequency used for the A/D conversion, internal digital processing and digital output will be generated internally. The sampling frequency (32KHz, 44.1KHz or 48KHz) can be selected with the button in the input section and is displayed by the lighting of a green LED in the d02's front panel sampling frequency display. For applications where analog input is used, but where the AES/EBU digital output of the unit must be synchronised with another AES/EBU digital audio signal or with a Word Clock signal, the input selector must be set to ANALOG extern.

External synchronization

The AES/EBU signal or Word Clock signal which the unit is required to synchronise the sampling frequency of its AES/EBU output with can be applied to the AES/EBU input connector or to the EXT SYNC word clock input connector respectively. The sampling frequency of the external AES/EBU or Word Clock signal (and hence the operating sampling frequency of the d02) will be indicated by the lighting of a yellow LED in the d02's front panel sampling frequency display.

All three LED's in the d02's front panel sampling frequency display will flash red if the d02 can not synchronise to an external sampling frequency signal because no signal is connected or because the connected signal is outside the admissable working sampling frequency range of the unit (30KHz to 50KHz).

Digital signal processor

The digital audio signal (either an AES/EBU digital input signal or an analog input converted by the A/D converter) is processed in a Texas Instruments Floating Point Signal processor with a data width of 32 bits. The use of 32 bit digital audio sample length in calculation ensures that there is no deterioration in signal quality, even if AES/EBU digital audio data with the maximum word length of 24 bits is input into the unit.

The DSP carries out the functions of the dynamics processing, the linear gain and the emphasis filtering. It measures the input and output levels and generates data for GAIN CHANGE display. Reading of the front panel buttons and operation of the front panel display is performed via a special interface (see also chapter 3.).

compression

One main task of the digital transmission processor d 02 is the **compression** of low signal levels. The compression- RATIO expresses the effects of a change of the input signal in dB on the change of the output signal in dB.
E.g. a ratio of 2:1 means that a change in input signal of 20 dB causes a change in output signal of 10 dB. With the choice of a compression ratio, the intensity of the compression is determined and with it also a certain compression characteristic (see also fig.2 and fig. 3). The RATIO parameter is adjusted on the front panel in four steps, from 1.1:1 to 2.0:1. The transition to another characteristic can be carried out during the running programme. It does not cause any clicking noises.

The lower the signal level, the higher the gain of the compressor will be. Independently of compression ratio, the maximum amount of compression gain can be adjusted so that no inadmissible increase of background noises (e.g. live atmos, air-conditioning, hum and noise) may occur during signal pauses.

Maximum compression gain

To set the maximum compression gain press both the RATIO button and the PROGRAM button simultaneously. A **red** LED becomes visible In the compressor gain display , which indicates the **maximum value of compression gain**. This value can be changed in 1dB steps over the range +2dB to +15dB pressing the **INC** and **DEC** buttons.

expander

The **expander** becomes effective when the signal level falls below an adjustable expander threshold. It is possible to select four thresholds from -60 dB...-30 dB.
If the level falls below the threshold , the gain is steadily decreased up to -15 dB. The downward regulation of the expander is achieved just as quickly as the upward regulation of the compressor, thereby compensating the resulting increase in signal noise.

Look ahead limiter

For the dynamics functions, particularly the algorithm of the **limiter**, a **signal delay** of approx. 2 ms is built in. This delay makes it possible to arrange the algorithm of the limiter in such a way that the control mechanism is activated before maximum level is reached. Within the rise time of the signal the peak level is recognised and the maximum is calculated in such a way that full scale level is reached precisely without causing clipping.

The processing of digital audio signals in the signal processor requires a machine-specific format. Special interface circuits are therefore available to convert to standardised digital interface formats.

D/A-converter

Additionally, an analog output signal is available. A stereo - D/A converter with a resolution of 20 bits generates an analog signal with very high audio quality. This signal is fed to balanced output drivers.The gain of the balanced analog output driver circuit for each analog output can be adjusted on the rear panel, so that the maximum possible analog output level can be adjusted to be any value in the range from +6dBu to +22dBu.

(The maximum possible analog output level here is the analog output level when the output level meter shows 0dBFs full scale digital level and the D/A converter is being fed with a digital signal at 0dBFs - the maximum possible full scale level that can be represented digitally).

If the internal digital reference level is reduced to below 0dBFs then the maximum analog output level will be correspondingly reduced by the action of the limiter.

For example if the internal digital reference level is reduced to -9dBFs (9dB below the maximum possible level that can be represented digitally) then the actual maximum digital level that will be received by the D/A converter will be 9dB below the maximum possible digital level. In this case the range of adjustment for maximum analogue output level will be -9dBu to +13dBu.

The design of the electronically balanced analog output drivers is such that the output level is maintained even when driving an unbalanced load.

APPLICATION NOTES

5

5.1. Presets

It is possible to choose one of eight different control characteristics for the dynamics processor. Each of the four different sets of control characteristics provides ideal dynamics control for a different type of programme signal as follows:

stereo mode	2-channel-mode
1 - universal	5 - universal
2 - pop music	6 – pop music
3 - speech	7 - speech
4 - live	8 - live

Selecting a particular preset sets up the optimum parameters of the dynamics processor (attack and release times, threshold levels and interactions between the multiple signal dependent control circuits) for a particular kind of programme material.
For example, generally speaking, release times are longest when using the universal setting and shortest when using the live setting. (In order to understand the basic Multi-loop principle of the Jünger Audio dynamics processors please refer to chapter 1.2).
Fixed presets containing optimised parameters for different types of programme signal are used because, with the great number of parameters used and the interactions of parameters used in different stages of the multiloop system, changing of individual parameters by the user could cause problems.

5.2. Processing signals containing emphasis

If the audio signal was recorded with **emphasis**, the additional information of the digital input signals contains a definite emphasis-control-bit in the AES/EBU or SP/DIF format. This is sometimes the case in older recordings because it slightly improved the signal-to-noise ratio of currently used analog-digital converters. Similar to noise reduction methods in analog magnetic tape recording, the higher signal frequencies are raised prior to recording, and subsequently lowered in playback, causing a lowering of the higher frequency noise level.

If such a signal is compressed or limited in a dynamics processor, problems will occur as the peak levels for high frequencies do not represent the true values. The dynamics processor causes a change in peak levels which would, however, lead to a change in the treble content after passing through the external deemphasis filter.

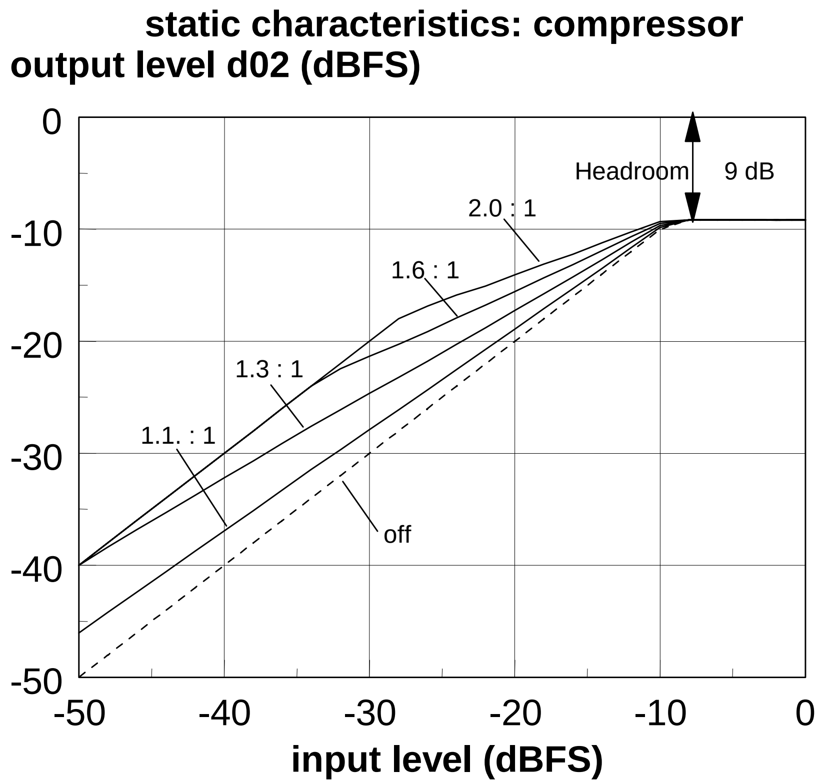
Prior to dynamic processing a signal recorded with emphasis must therefore be linearized, i.e. pass through a digital deemphasis filter. This filter in the **d 02** is automatically switched on if the corresponding control bit is set in the AES/EBU or S/PDIF format. If the filter is turned on, the colour of the AES input LED will change to red.

5.3. Working with headroom

The static characteristics of the **d 02** (see also fig. 2) usually refers to the **digital reference level 0 dBFS** (dB Full Scale). This is useful for most applications of the dynamics processor as the on-following digital recording system is supposed to be balanced down to the final bit.
For applications using headroom the **d 02** can be adjusted to another reference level of **0 ... -15 dBFS** in steps of 1 dB. The limiter threshold and therefore the maximum output level are determined by this digital reference level. This value is then also the reference for the expander and limiter threshold values. The static characteristics for a reference level of -9 dBFS are illustrated in fig. 5.

The adjustment of the device to this reference level is achieved with pushing DISPLAY and GAIN buttons at the same time (see also chapter 2.4. and 3.).

fig. 5: Static characteristics: Compressor/ Limiter with -9dBFS Digital Reference Level



compression gain: max. 10 dB
parameter: ratio
digital reference level: -9 dBFS

5.4. Influence of signal delay time

The audio signal delay through the dynamics processor is approx. 2ms due to delaying of the audio signal using internal memory. A small delay is deliberately introduced to the audio signal in order to allow limiter and compressor algorithms which can 'preview' the audio signal before changing it. That is the signal curve can be changed before maximum level is reached. (For further details see chapter 1).
This delay must be considered before attempting to mix signals processed by the dynamics processor with other undelayed signals.

When mixing together a delayed signal and a direct signal there may be cancellation of the signal waveform at some frequencies and re-inforcement of the waveform at other frequencies (comb filter effect). Corresponding 2ms delay of direct signals should therefore be carried out before mixing them with delayed processed signals.

5.5. Selection of parameters to increase loudness

Signal compression and the loudness enhancement of the digital audio signal resulting from it can be achieved by combining two dynamic range control processes: firstly, the **compression** achieved by increasing small and medium signal levels and secondly, **linear amplification** combined with the inaudible **limitation** of individual, remaining peak levels with the limiter.

In the gain change mode the operation of compressor and limiter can be observed on the display. For smaller signal levels the compressor causes additional amplification which however decreases the higher the signal level is . With full scale levels the compressor is practically ineffective so that even an increase of the RATIO will have no effect.
If you now increase the linear amplification GAIN, individual peak levels are raised above the limiter threshold and limited inaudibly. All other signal components can however be increased. If the gain is too large also medium levels are treated by the limiter, which means that the limiter then reduces the signals continually and again reduces the additionally applied amplification.
The display for Limiter-Gain-Reduction should be in the region of 0...-6...-8 dB and should not light up red continuously, so that a dynamic limitation only applies to signal peaks. Then the signal compression and therefore also the increase of loudness is at its most effective.

5.6. Redithering - Reduction of word length of digital output signal

At the end of postproduction the material must be prepared for copy on COMPACT DISC. The information of a 24 bits signal is not more storable linear. One must shorten 24 bits data word to 16 bit word length. The practice offers several procedures for it.

truncation

In the simplest case, the last bits are cut off - truncation. One requires no further processing to this, it is enough to record a 20or 24 bits signal direct on a 16 bits storage medium. In this case, a not unimportant quantization mistake however results, the part of the harmonious distortions increases. Single numeric roundoff of the signal to 16 bits reduces this mistake. Nevertheless, the result will normally be worse than the data by the same original analog signal converted with a good 16 bit ADC.

redithering

In order to receive a better quality during cut down the data to 16 bit one must redithering the material with corresponding devices. Here the device is calculating random numbers (dither signal) and add a different random number to every sample. Then it will be cut off to 16 bit. As a result, the bit with least weight (LSB) is put in such a way that it corresponds best to the information of the last bits following available ones no more and makes less distortions as hissing in the signal.

noise shaping

A specific redithering is noise shaping, with that the noise modulation of the LSB is considering the psycho-acoustic sensitiveness of the human ear. With it can be enlarged the audible dynamic range of a reformatted 16 bit recording ideally by about 3 bits i.e. to more than 110 dB. To make this quality audible, the CD must of course be played back with appropriate monitoring equipment (D/A converter, amplifier) for 20 bit quality.

Disadvantages of noise shaping/ redithering

The disadvantage of redithering - noise shaping consists in the restrictions during postproduction. In order to receive the achieved effect every processing must occur with coefficients which correspond to the word length of the initial data. At signals processed with noise shaping, these coefficients would have to be besides filtered in the same manner like the dither signal. If one can not adhere to these conditions, one must live with the loss of the effect, in some cases it occurs data losses. Multiple application of these procedures can even make drop-in and noises, such as twittering or clicks. Therefore, noise shaping or redithering should be used only at the end of the process chain, i.e. during the preparation of the copy master for reproduction.

APPLICATIONS

6

- **mastering** of **CD, DCC, MD**
  maximum recording level without clipping
  increased programme density and loudness

- digital **recording and mixing**
  increased loudness level (compressor, limiter)
  eliminating noise signals (expander)

- **FM-Broadcast, TV-Sound**
  signal conditioning
  matching dynamic range of different programme signals
  increasing signal loudness level

- limiter for **digital or analog transmission links**
  always digital full scale signal, without clipping

- **post production and ADR studios**
  adjusting dynamic range and loudness level of individual takes,
  maximum recording level without clipping

- **A/D converter free of overload** for general use
  high performance 24 bit ADC in combination with digital limiter
  digital output signal without clipping

further applications without the dynamic functions

- digital audio **format conversion**
  all digital outputs are available in parallel
  irrespective of the input format
  AES/EBU + S/PDIF + OPTICAL

- **digital deemphasis filter**
  removing emphasis automatically
  emphasis bit in AES/EBU is also removed

- **digital-analog converter**
  high quality 24-bit stereo output signal
  balanced line outputs with adjustable output level

TECHNICAL SPECIFICATION

7

digital input / output

sample rate :	30 kHz ... 50 kHz
audio data format :	24-bit (AES/EBU)
	24-bit (A/D-,D/A-converter)

AES/EBU

level :	5 Vpp / 110 Ohm, balanced
connector :	XLR
input format :	AES professional, AES consumer
output format :	same as input

S/PDIF

level :	0.5 Vpp / 75 Ohm, unbalanced
connector :	RCA
input format :	AES professional, AES consumer
output format :	same as input

OPTICAL

connector :	TOSLINK

analogue input / output

A/D-converter :	stereo, 24 Bit, oversampling
dynamic range :	112 dB (RMS)
	114 dB (A-weighted)
input level :	+12...+22 dBu for 0 dBFS, adjustable
input :	XLR, floating balanced, 10 kOhm
	(optional: transformer balanced)

D/A converter :	stereo, 24-bit, oversampling
dynamic range :	108 dB (RMS)
	110 dB (A-weighted)
output level :	+12...+22 dBu for 0 dBFS, adjustable
output :	XLR, floating balanced, 50 Ohm
	(optional: transformer balanced)

general

remote :	for connection with d - remote drc01
	(optional)
power consumption :	approx. 20 W
dimensions :	19 inch, 1 RU, 250 mm depth
weight :	appr. 4.5 kg

8

WARRANTY AND SERVICE INFORMATION

JÜNGER AUDIO grants a two-year warranty on the

d02 dynamic range processor

If the unit has to be serviced, please send it,
ideally in the original box, to:

JÜNGER AUDIO - Studiotechnik GmbH

Justus-von-Liebig-Str. 7

D - 12489 Berlin
GERMANY

Tel.: (*49) -30-677721-0
Fax.: (*49) -30-677721-46

KONFORMITÄTSERKLÄRUNG
DECLARATION OF CONFORMITY

Geräteart :	**Digitaler Dynamikprocessor**
Type of equipment :	**digital dynamics processor**

Produkt / Product : **d02**

Das bezeichnete Produkt stimmt mit den Vorschriften folgender EU-Richtlinie(n) überein:
The aforementioned product complies with the following Europaen Council Directive(s):

89/336/EWG
(geändert durch 91/263/EWG und 92/31/EWG)
(changed by 91/263/EEC and 92/31/EEC)
Richtlinie der Rates zur Angleichung der Rechtsvorschriften der Mitgliedsstaaten über die elektromagnetische Verträglichkeit
Council Directive on the approximation of the laws of the Member States relating to electromagnetic compatibility

73/23/EWG
(geändert durch 93/68/EWG)
(changed by 93/68/EEC)
Richtlinie des Rates vom 19. Februar 1973 betreffend elektrische Betriebsmittel zur Verwendung innerhalb bestimmter Spannungsgrenzen
Council Directive of February 19th 1973 concerning electircal equipment for operation within certain voltage limits

Zur vollständigen Einhaltung dieser Richtlinie(n) wurden folgende Normen herangezogen:
To fully comply with this(these) Directive(s), the following standards have been used:

EN 55022 :	1987
EN 50082-1 :	1993
EN 60065 :	2002

Dieser Erklärung liegen zugrunde :	Prüfbericht(e) des EMV-Prüflabors Interne Vorschriften zur Sicherheits-Prüfung
This certification is based on :	Test report(s) generated by EMC-test laboratory Internal regulations for safety check
MEB Messelektronik Berlin :	Kalibrier- und Prüflabor accredited EMC laboratory
Aussteller / Holder of certificate :	Jünger Audio Studiotechnik GmbH Justus-von-Liebig-Strasse 7 D - 12489 Berlin

Berlin,	18.03.2003	....
(Ort/Place)	(Datum/Date)	(Herbert Jünger, Geschäftsführer/Managing Director)

d02

Jünger Audio GmbH
Justus-von-Liebig-Straße 7
12489 Berlin
Germany

phone: +49 30 6777 21 0
fax: +49 30 6777 21 46
info@jungeraudio.com
www.jungeraudio.com