OPERATIONS MANUAL

Digital Dynamic Range Processors

d01

d02

digital dynamics processor

d01

release 4.0

jünger audio

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INTRODUCTION

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The **digital dynamics processor model d 01** is a professional studio device that processes the dynamic range of digital , as well as analog audio signals.

The **Digital Dynamics Processor model d 01** is a professional studio device that processes the dynamic range of digital audio signals.

With the help of the **limiter** and the **compressor** it is possible to achieve the highest possible digital full scale level without clipping. The increases in programme density and loudness are entirely free from the processing noises typical of dynamic range processors such as "pumping" and "breathing", or signal discolouration.

The unit is easy to operate and requires only a limited number of manual settings to be made by the user to achieve optimum results. 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
  Sony SDIF-2
  input and output may be different
- **analog output**
  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 01 digital dynamics processor** can be used to process digital 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 S/PDIF) and in the SDIF-2.
Input and output can be selected independently. The output signals are available in parallel in all 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 01 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).

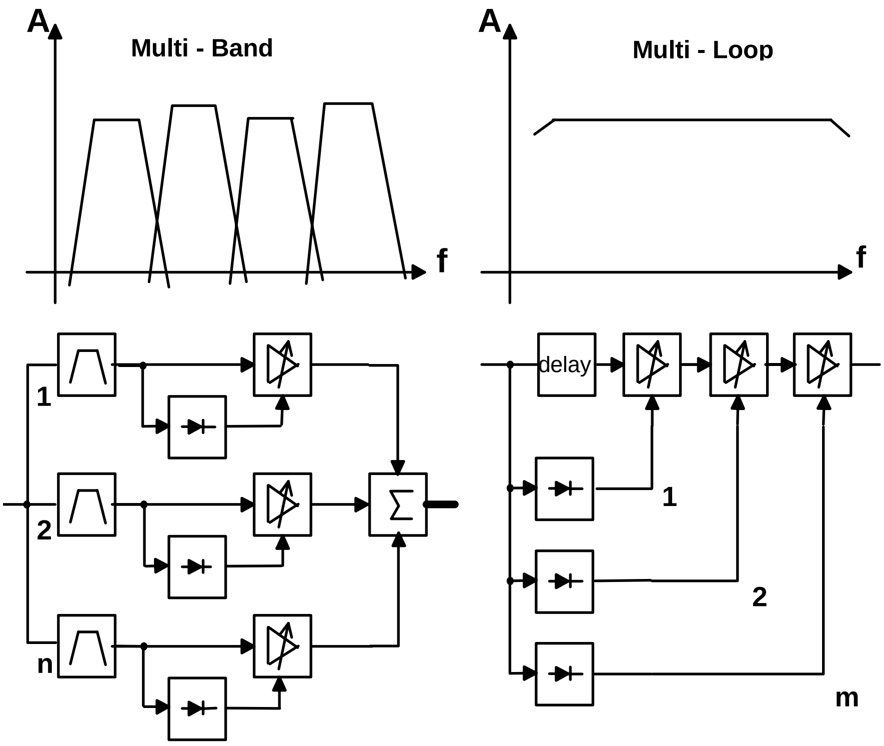


fig. 1:

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.

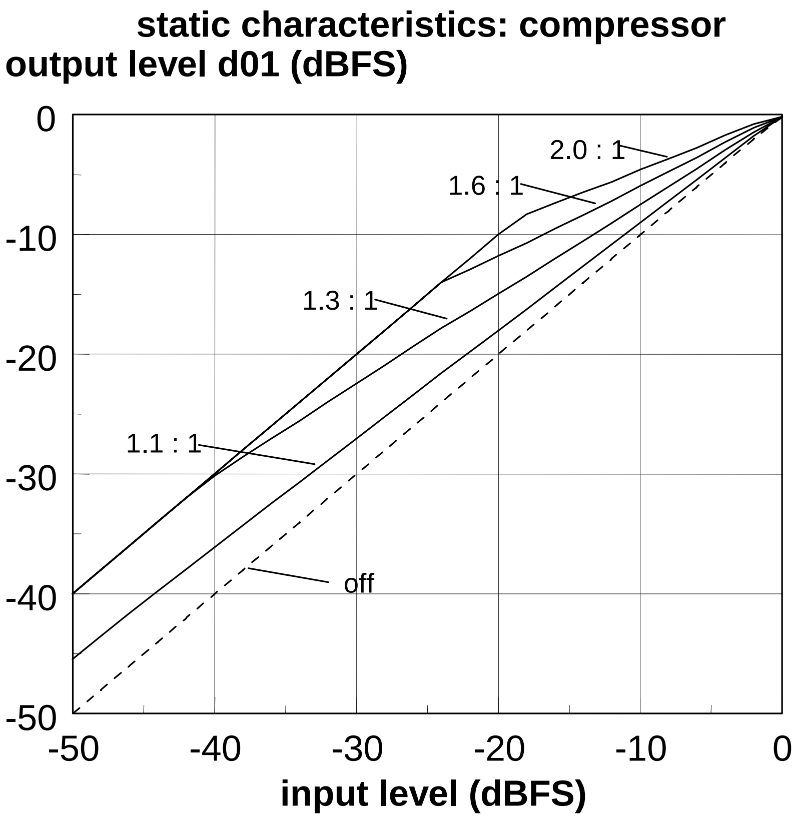
compression gain

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

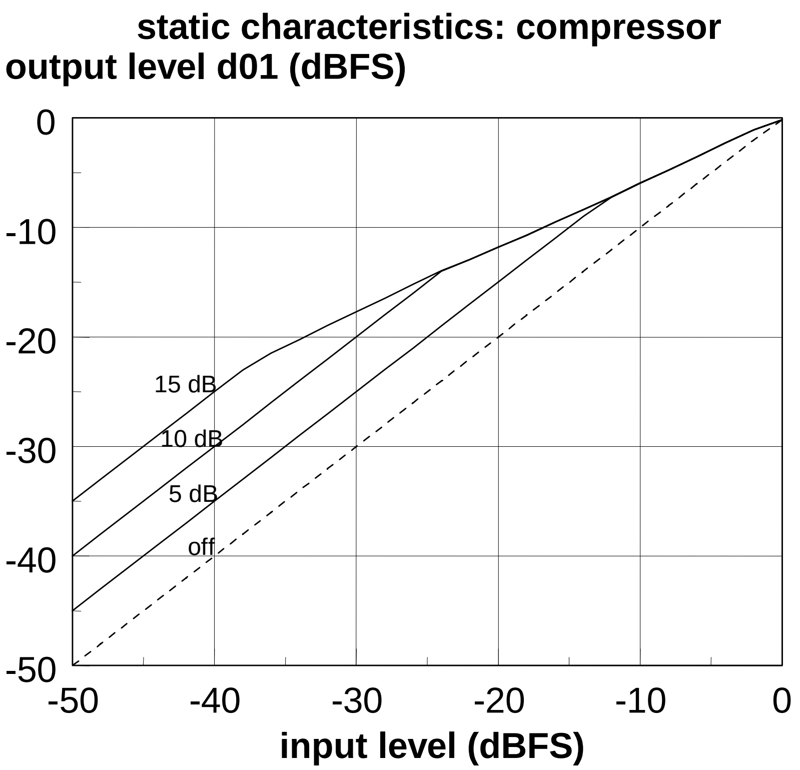
Below an adjustable threshold level an **expander** can be activated which can lower the amout of noise signals.

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

INSTALLATION

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2.1. Power Supply

The digital dynamics processor d01 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 d01 are arranged in functional groups on the rear panel.

POWER INPUT

IEC mains input connector 230 V, 50 Hz (UK: 240 V, 50 Hz; JAPAN: 100 V, 60 Hz; USA: 127 V, 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

SDIF-2

Sony Digital Audio Format
Different lines for SYNC (Wordclock), CH 1 and CH 2
Input and Output : BNC, 75 Ohm

DEEMPH

Switch for manually setting the deemphasis filter. If the filter is switched on Input LED is lighting red.

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. Setting the Digital Reference Level

The static characteristics of the processor d 01 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.

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

CONTROL AND DISPLAY ELEMENTS

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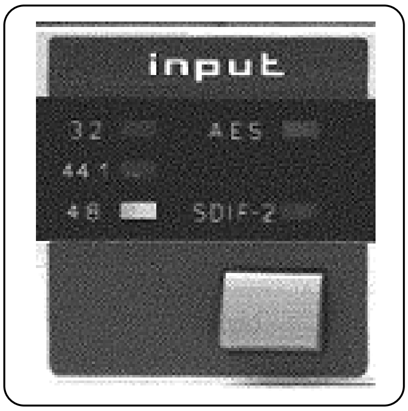
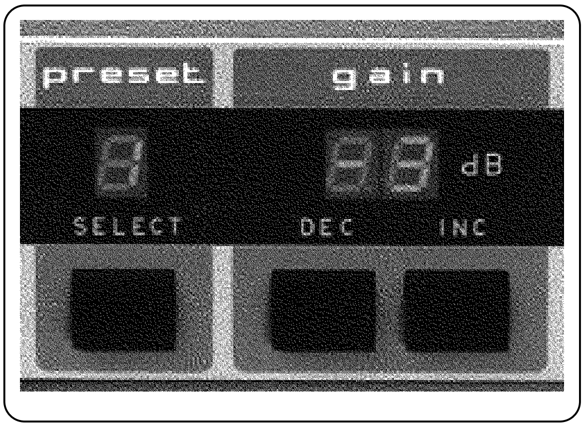
All functions of the **digital dynamics processor d 01** are activated by buttons. The front panel shows easily recognizable function groups.



input

By pressing the 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 **green** to show the newly selected input (AES or SDIF-2).. If the LED lits **red** the input signal was recorded with **emphasis** or the deemphasis switch on the rear panel is on.
(see also chapters 2.2. and 4.2.).
To the left 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.

Following the dynamic range processing,the digital audio signal selected is available at all outputs in parallel, i.e. the digital format of the output signal can be selected irrespective of the input format.



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.

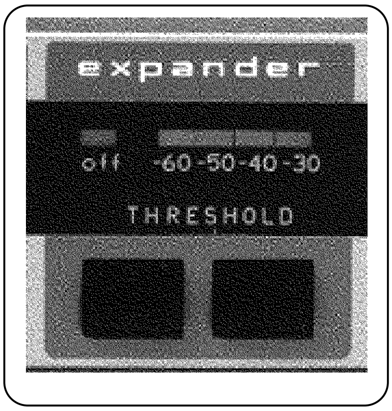
in stereo mode (loop function)	in 2-channel mode (loop function)
1 - universal	5 - universal
2 - popl music	6 - pop music
3 - speech	7 - speech
4 - live	8 - live

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

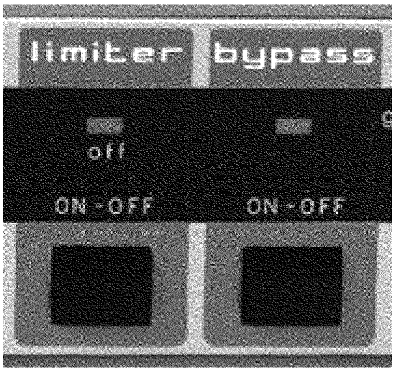
The expander **THRESHOLD** can be changed upward or downward with two buttons and is visible on the LEDs above them. Four expander thresholds (-60 dB, -50 dB, -40 dB, -30 dB) can be selected. The threshold level is related to the choosed digital reference level.
In the OFF position the expander function 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.

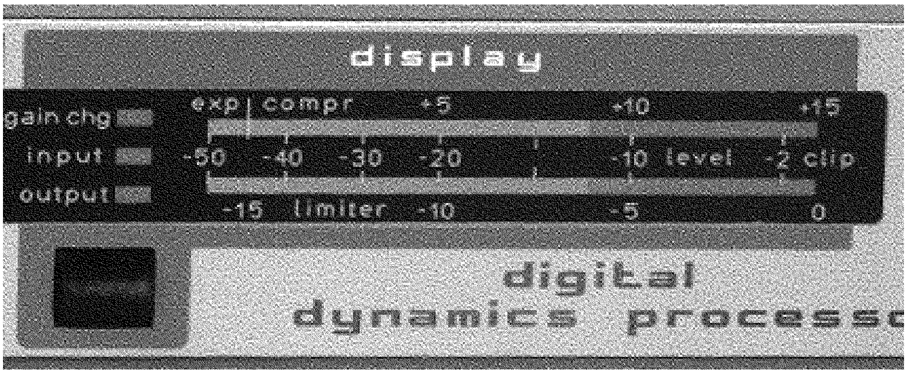


limiter

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

4

Power-on Setting

After switching the power on, the **digital dynamics processor d01** automatically chooses the settings used before the power was turned off.
All parameters used, e.g. input, preset, gain, expander, compressor 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 in the three most commonly used signals (AES/EBU, SDIF-2). 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 d01 directly measures the actual sampling frequency of the 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 d01 can synchronise to properly.

Digital input signals - AES/EBU

If the d01 can not synchronise properly to a supplied 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 d01 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 d01'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 d01'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.

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.

The processing of data for the Sony **SDIF-2** formats is done in special interface circuits. The Sony format contains additional data that must be separated from the pure audio data and rejoined with the output signal after processing. In the SDIF-2 format the emphasis bit is also decoded and used for the automatic control of the deemphasis filter.

With the digital **input** selection button on the front panel the propriate input is selected. At the same time, the entire signal processing, including all digital outputs, is synchronized to this input.

Digital signal processor

The digital audio signal 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 d01 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 classical setting and shortest when using the speech 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.

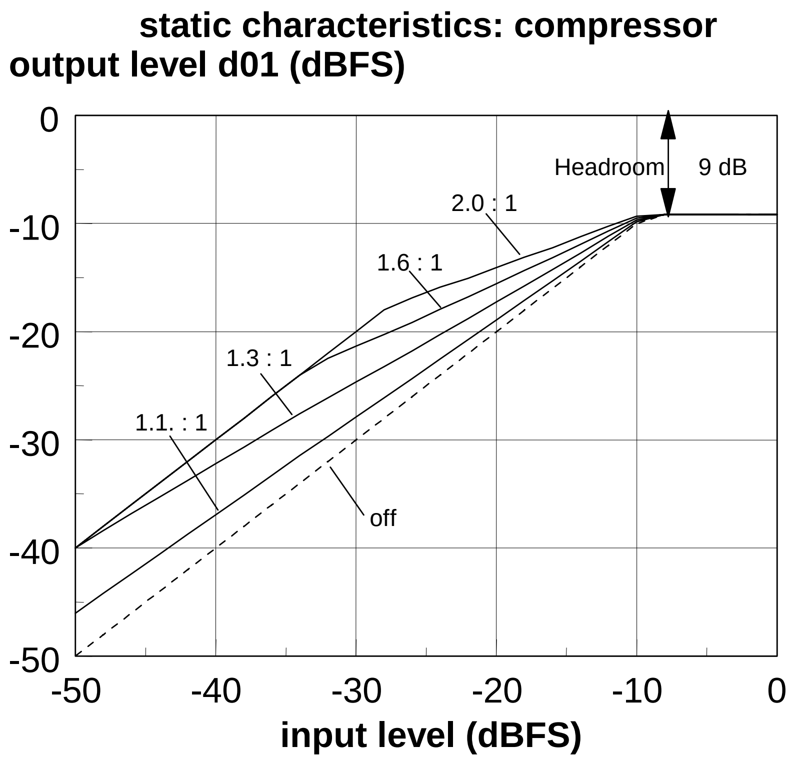
Prior to dynamic processing a signal recorded with emphasis must therefore be linearized, i.e. pass through a digital deemphasis filter. This filter in the **d 01** is automatically switched on if the corresponding control bit is set in the AES/EBU or S/PDIF format. If the filter is turned on, the colour of the AES input LED will change to red.

5.3.
Working with headroom

The static characteristics of the **d 01** (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 01** 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.

APPLICATIONS

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- **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

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 + SDIF-2

- **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

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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

SDIF-2

level :	TTL, 75 Ohm
connector :	BNC, 75 Ohm
format :	SONY SDIF-2

analogue output

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

WARRANTY AND SERVICE INFORMATION

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JÜNGER AUDIO grants a two-year warranty on the

digital dynamics processor MODEL **d01**

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: Digital Filter Processor
Type of equipment: digital filter processor

Produkt / Product: **model d01**

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/EWG and 92/31/EWG)
Richtlinie der Rates zur Angleichung der Rechtsvorschriften der Mitgliedsstaaten über die elektromagnetische Verträglichkeit
Council Directive 89/336/EC on the approximation of the laws of the Member States relating to electromagnetic compatibility

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

Dieser Erklärung liegt zugrunde: Prüfbericht(e) des EMV-Prüflabors
This certification is based on: Test report(s) generated by EMC-test laboratory

MEB Messelektronik Berlin — Kalibrier- und Prüflabor
accredited EMC laboratory

Aussteller / Holder of certificate: Jünger Audio Studiotechnik GmbH
Rudower Chaussee 5 (IGZ)
D - 12489 Berlin

Berlin,	02.11.1995	....
(Ort/Place)	(Datum/Date)	(Rechtsgültige Unterschrift/Legally Binding)

professional audio products

digital dynamics processors	d01, d02 accent1, accent2
digital filter processor	e07
surround products	multichannel digital dynamics processor ORION 5.1 level controller 206
digital voice processing	voice and monitor processor v01 digital voice processor v02 dual channel voice processor v03 digital voice processor v05
digital desktop mixer	mix4
transmission signal processing	digital transmission processor d07 digital transmission limiter mpx01
4channel processors b40series	digital audio toolbox b40 digital audio limiter b41 digital dynamics processor b42 digital audio toolbox b43 SDI audio converter / router b44 digital audio delay b45
digital audio modular processing system C8000	
SDI interface modules SDI20	

jünger audio

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