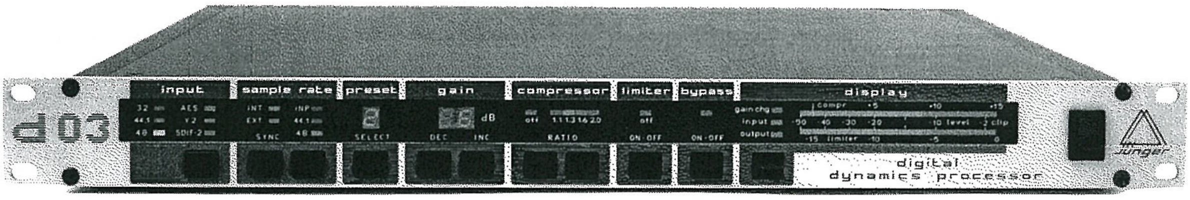
DIGITAL DYNAMICS PROCESSOR

model d03



operation manual

rev. 2

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CONTENTS

The **digital dynamics processor d 03** is a professional studio device that processes the dynamic range of digital audio signals.
The unit contains a digital compressor and limiter and the capability of **sample rate conversion**.
Input and Output may operate with different sampling rates or with the same sampling rate but do not have to be synchronised.
The dynamics processor d 03 has multiple digital interfaces (AES, SP/DIF, OPTICAL, YAMAHA Y2, SDIF-2) and can also be used as a format converter.
Analog inputs and outputs are not available.

With the help of the **limiter** and the **compressor** it is possible to achieve the highest digital full scale signal without clipping. An increase in programme density and loudness level is 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.

Features

- fully **digital** processing device (stereo)
  audio data word length: **24 bit**

- **compressor, limiter**

- **4 presets** (universal, classical, pop music, speech) for stereo or 2-channel mode
  complex, programme-dependent control algorithms

- linear **gain**
  - 6 dB ... +15 dB, in 1 dB steps

- **sample rate conversion**
  independent of dynamics

- digital **deemphasis filter**
  automatic or manual

- multicoloured **LED display**
  shows either input level, output level or gain change
  with peak hold and digital full scale display

- **multiple digital audio interfaces**
  AES/EBU + SP/DIF + OPTICAL
  Yamaha Y2
  Sony SDIF - 2

1. THE DESIGN OF THE DEVICE

The **digital dynamics processor d 03** was designed to process digital audio signals. It can be connected to all common digital audio formats.
Apart from the **AES/EBU** standard, including SP/DIF and OPTICAL, signals of the YAMAHA **Y2** and Sony **SDIF-2** formats can also be processed. The output signals are available in parallel in all the digital formats so that, depending on the active input, a format conversion can also be carried out.

The digital sample rate converter of the d 03 processor works independently of the dynamics. This feature allows free selection of the output sample rate.
The sample rate of the output can be internaly generated by crystal oscillators (44.1 or 48 kHz) or can be synchronized to an external wordclock or to the input signal.

The increase in signal density which makes an increase in the loudness of the digital audio signals possible, can be achieved by the interaction of two dynamic range control processes;
first by the **compression** of low and medium signal levels and
second 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).

1.1 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 mudulation 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.

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 program signal and must be changed for other signals.

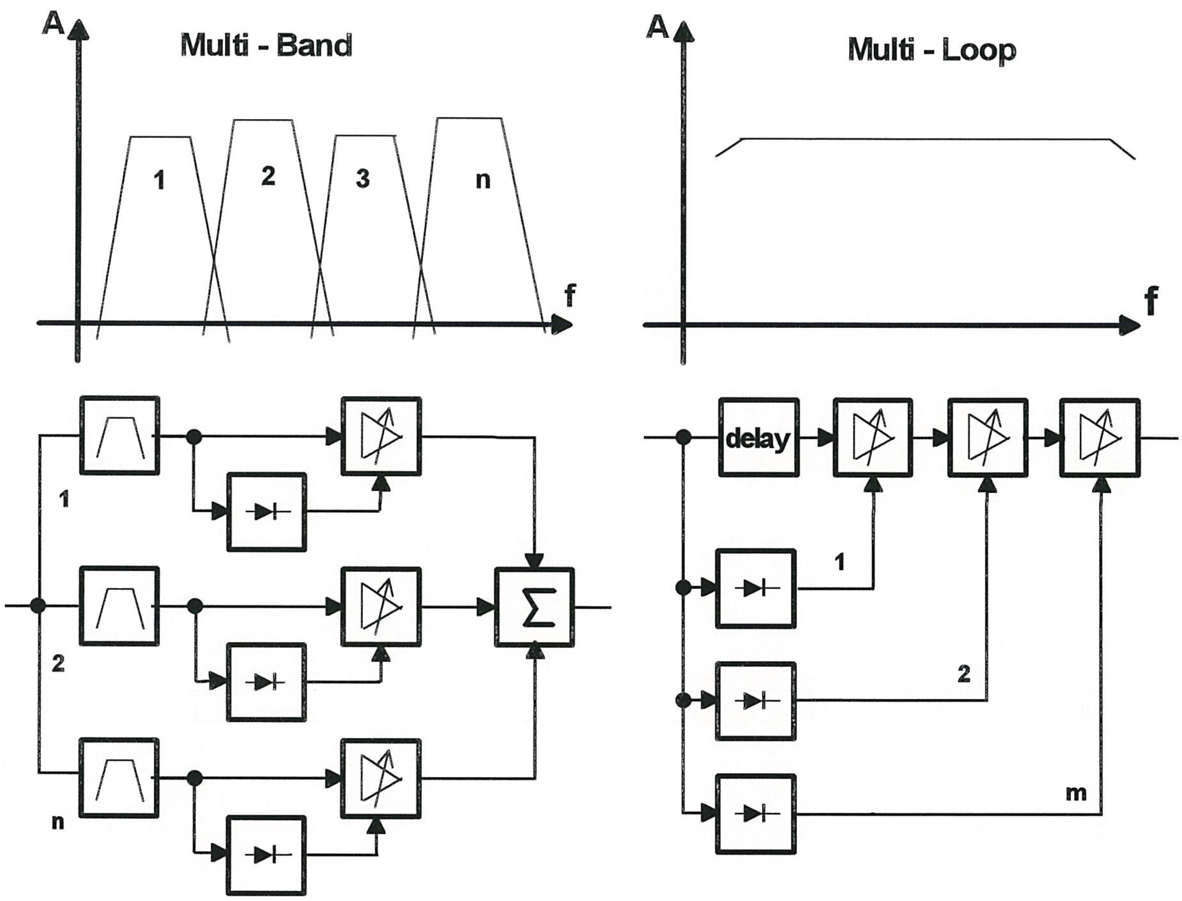
Dynamic range processors which split the spectrum into a plurality of bands, i.e. which have a multi-band structure, have some advantages over traditional compressor designs. The dynamic control parameters in each band are independent of one another and can be set in such a way that a broad program range 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 the 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.
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 are 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.

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 impotant 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.

Figure 1 shows the basic principles of dynamic range processors.



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 03 digital dynamics processor** is a dynamic range processor which, contrary to its conventional counterparts, is effective for a wide dynamic range of the input signals (50 dB). 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 sound impression.

The lower the signal level, the higher the gain of the compressor will be. 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 atmo, air-conditioning, hum and noise).

The static compression characteristics of the d 03 compressor are shown in fig. 2 and fig. 3.

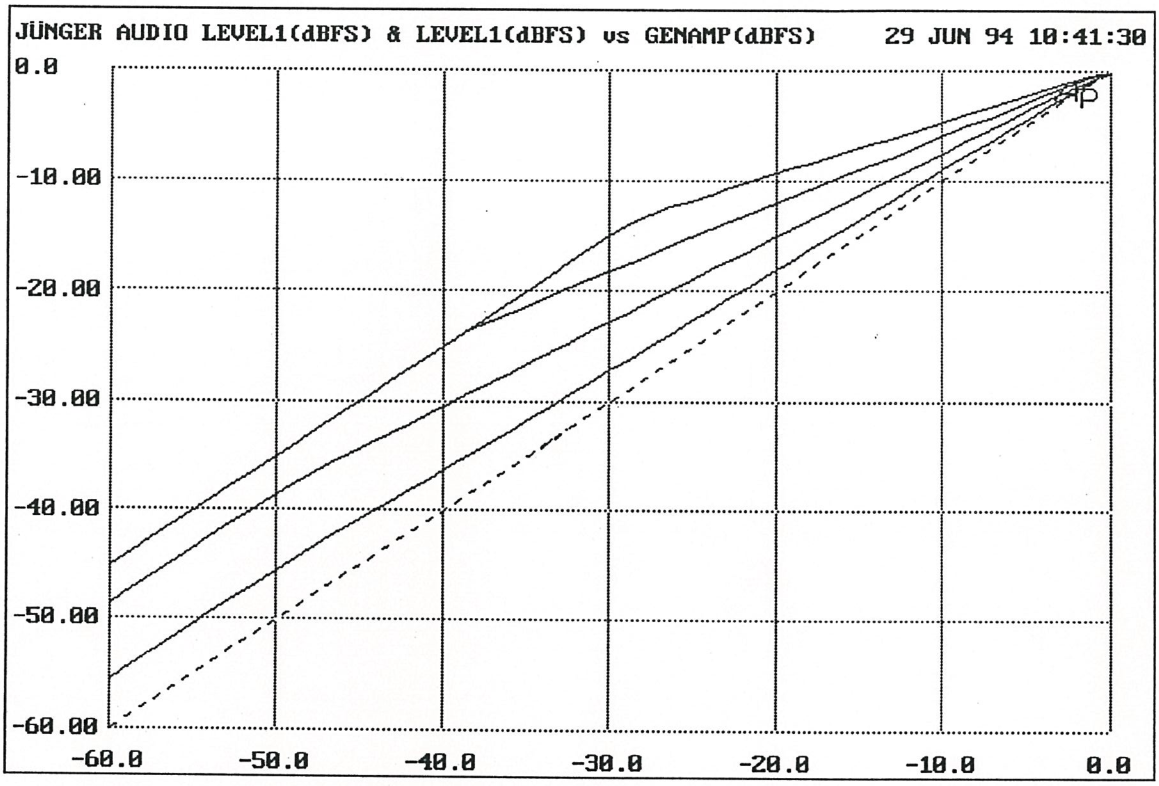
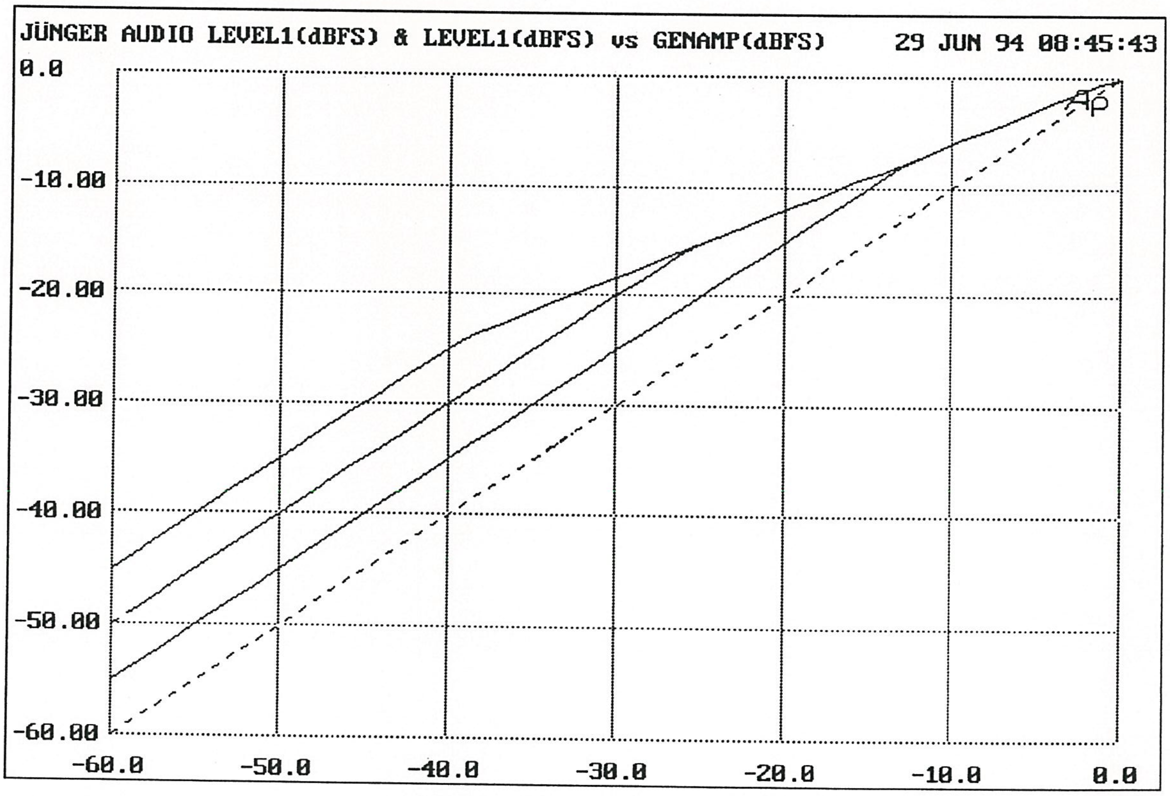


Fig.2

static characteristics, compressor
compression gain: 15 dB
parameter: RATIO



static characteristics, compressor
RATIO: 1.6
parameter: compression gain

Fig.3

1.2. Sample Rate Conversion

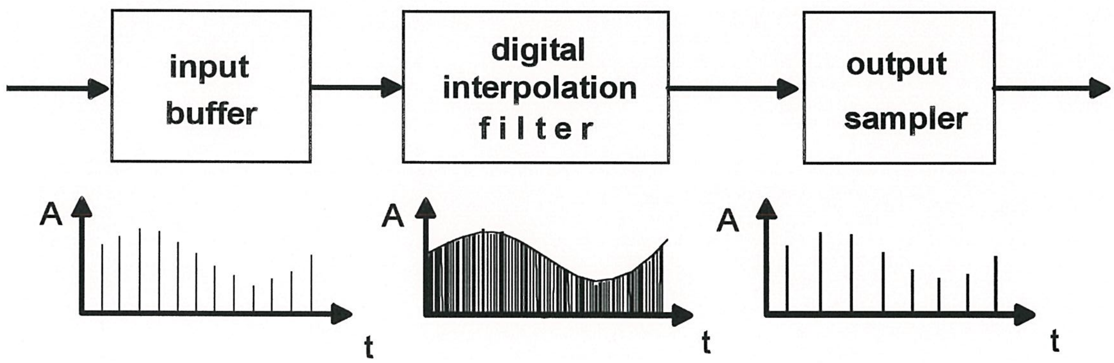
The sample rate conversion feature of the d 03 digital dynamics processor can be used independently of the dynamic range functions. The ability to separate the sample rate of input and output allows the input signal to be used like an analog signal.
Therefore it is not necessary to synchronize to the input, which means that the input signal may run free.
The output sample rate can be referred to the crystal oscillator frequencies 44.1 kHz or 48 kHz or can be determined by external word clock signals.

Normally the sample rate converter is used to change the sample rate (e.g. Input DAT-recorder 48 kHz, output U-Matic 44.1 kHz) but can also be used when input and output have the same clock rate. This application is very effective to remove clock jitter from the input signal (reclocking).

The process used in the d 03 for changing the sample rate is called **asynchronous sample rate conversion** because there does not have to be a relationship between input and output sample rate. The principle on which it functions is similar to the method of sample rate conversion to go back to analog signals but without the disadvantages of the analog method.

The input signal consists of discrete sample values which come in with a temporal distance depending on the input sample rate. A digital-analog converter followed by a low pass filter generates a continuously analog signal without temporal gaps. This continuous signal can be converted back to digital again with an analog-digital converter . The output sample rate is independent of the input sample rate.

Transforming this method into the digital domain means recovering a continously digital signal without temporal gaps from the incoming sample values. This is done with a digital interpolation filter making continuous digital signal values available at any time.
The signal then can be sampled again with the output rate.

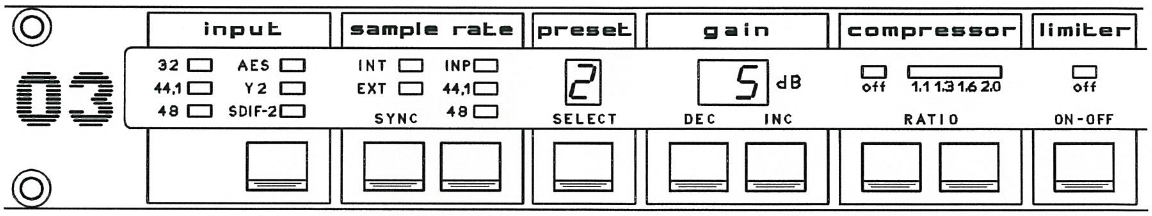


The principle of asynchronous sample rate conversion is shown in fig. 4.

2. OPERATING CONTROLS AND CONNECTIONS

2.1. Front Panel

All functions of the digital dynamics processor d 03 are activated by buttons. The front panel shows easily recognizable function groups.



input

By operating these buttons the appropriate digital input is selected and the signal is switched to the processor. Renewed operating of the button will lead to the selection of the next digital input in order. The input selected (AES, YAMAHA Y2 or SDIF-2) is shown on the LED display, normally with a **green** light. If the LED shows a **red** light, this means that either the input signal was recorded with **emphasis** or that the deemphasis switch on the rear panel is on .

To the left of the digital input indicator are three LEDs which shows the **sample rate of the selected input.** If a given signal 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 input sample rate is outside the admissible tolerance range.

sample rate

With the function buttons in this field the **sample rate of the output signal** can be determined. Internal (INT) or external (EXT) synchronisation is selected by pushing the **SYNC** button . The sync-mode selected is shown with a corresponding yellow LED. Internal synchronisation means that the sample rate of the output signal is determined by reference clocks inside the device. With external synchronisation the sample rate is locked to the external word clock applied at the rear panel.

The source of synchronisation can be selected in the internal synchronisation mode with the button on the right
When the input is selected, the coresponding yellow LED **INP** lights up and the sample rate of the output signal is equal to that of the input signal. In this mode the sample rate converter is without a function and the output rate is phase-locked with the input rate (sample rate bypass).

The processor d 03 can also be a master. In this case the sample rate is derived from internal crystal oscillators with a **44.1** or **48** kHz word clock. A **green** LED will show the selected frequency.

In the external synchronisation mode the same LED shows the frequency of an external word clock signal, but in **yellow** colour to distinguish it from the internal synchronisation mode. All LEDs will blink **red** if the external word clock signal is lacking or the rate is outside the admissible tolerance range.

preset

Press the SELECT button to choose one of the four operating programs of the unit which best corresponds to the kind of audio programme material which is being produced or transmitted. 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 - classical music	6 - classical music
3 - pop music	7 - pop music
4 - speech	8 - speech

To change the program loop is possible by the hold-down-function of the display button (see under display button).

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. Prolongued pressure on the buttons 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 (damping) being accidentally programmed.

compressor

The compression **RATIO** is adjusted with the appropriate buttons and is shown in the multi-coloured LED above. Four ratios can be selected (**1.1** : 1/ **1.3** : 1/ **1.6** : 1/ **2.0** : 1).
In the **off** position the compressor function is turned off.
The maximum amount of compression gain can be adjusted independently of the compression ratio . Both RATIO buttons have to be pressed at the same time. A red LED is visible in the compressor gain display which determines the **maximum value of compression gain**.
This value can be changed with the keys **INC** and **DEC** in the range of 2 dB ... 15 dB.

limiter

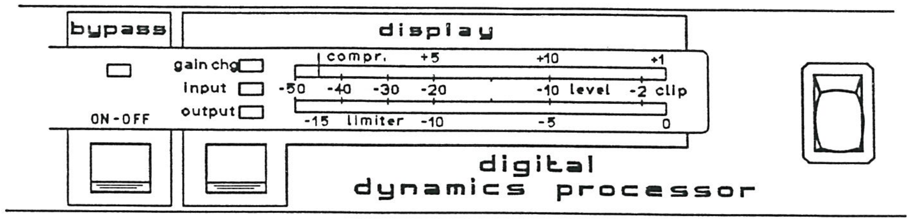
The limiter holds the digital output signal of the d 03 precisely to the set digital reference level. This function can be switched on and off with the appropriate button. As the limiter should always be activated in order to avoid an overload, the LED shows a **red** warning signal when the limiter is turned **off**.
The limiter works with a look ahead time (signal delay) of approx. 2 ms and already starts to reduce the signal before it has reached the maximum level. This delay time is present even when the limiter is turned off.

bypass

In the bypass mode (corresponding LED lights up **red**) the digital signal is fed through unprocessed directly to the output. To avoid phase jitter of the audio signal when turning the bypass on and off, the signal delay time of approx. 2 ms is also active in bypass mode.

The bypass function is not a relay bypass which includes all inputs and outputs and is therefore not effective when the device is turned off from the mains power.

display



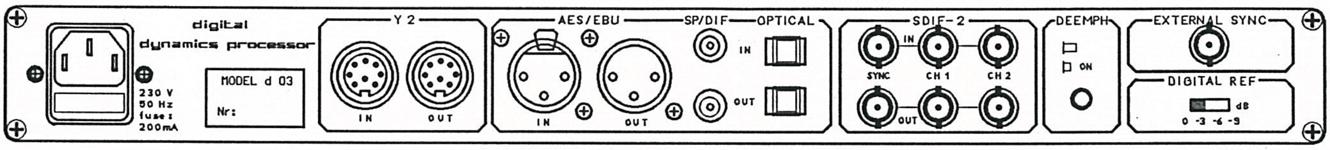
The two-channel LED display has three display modes (**input** level, **output** level and **gain change**). Each display mode can be selected with one button and is indicated with different coloured LEDs on top of that button. For better visibility each display is shown in its own colour.

Green shows the **input level** and **yellow** the **output level**. The scale located between the two bars indicates the levels. The display ranges from -50 ... 0 dBFS (dB Full Scale) and 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 existence and the level of digital input and output signals. A **peak hold** function is available for input and output which makes an improved registration of a momentary peak level possible.
The level display also shows overloads of digital full scale, when upper level **clip** LED lights up **red**. The clip indicator in the input mode makes the depiction of digital overload possible which is already present in the input signal. When using the limiter, the output signal is always free of overloads.
The level display is digital without integration time and processes **every** sample value.
The third display mode **gain change** shows the current control levels of limiter and compressor **in dB**. They are shown on the scales above and below the display bars. As the compressor inserts additional gain (i.e. no gain reduction), the direction of the display is opposite to that of the limiter.
A red LED is visible in the compressor gain display , which determines the **maximum value of compression gain**. This value can be changed with the buttons **INC** and **DEC** in the range of 2 dB ... 15 dB.

The DISPLAY button has a second function in addition to changing the display mode. It is used for setting the **stereo/2-channel mode** and 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 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 for the PRESET button.
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.

2.2. Rear Panel

All possible connectors of the **digital dynamics processor d 03** are arranged in functional groups on the rear panel.



Power Input
Power plug for 230 V, 50 Hz power supply with integrated fuse

Y2
Input and output for the YAMAHA Y2 digital audio format
8-pin. Connector corresponds to DIN 45326

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

SP/DIF
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 (do not use together with SP/DIF)
Input and output : TOSLINK

SDIF - 2
Sony digital audio format, SYNC (wordclock), CH 1 and CH 2
Input and output : BNC, 75 Ohm

DEEMP
Switch permits manual activation of the digital deemphasis filter

EXT SYNC
Word clock input for external synchronisation
Input : BNC

DIG REF
Selection of the digital reference level

3. FUNCTIONAL DESCRIPTION

When the power is switched on, **the digital dynamics processor d 03** automatically chooses the settings used before the power was turned off.

All parameters used, e.g. input, preset, gain, compressor and display, are stored and reapplied. The only exception is the limiter which, as a safety function, is always activated when the power is switched on.

The block diagram of the signal processing in the **d 03** is shown in fig. 5.

The device is capable of processing digital audio signals in the three most commonly used formats (AES/EBU, SDIF-2, Y2). The input signal is synchronized automatically and the sample rate may lie somewhere in the range from 30 kHz...50 kHz including all intermediate values ,
The sample rate is directly measured with a frequency counter and there is no decoding of definite control bits in the digital data stream.
For the standard frequencies (32 kHz, 44.1 kHz or 48 kHz) the corresponding LED will light up yellow on the front panel. This LED serves both as an indicator for a correct digital signal at the corresponding input and for a synchronization carried out properly. If no synchronization has taken place all three LEDs will flash red, which means there is no input signal or the sample rate is outside the admissible tolerance range.

Signals in the standard format **AES/EBU** run via the **XLR** connector and a balanced transformer to the AES interface. There the synchronization to the input sample rate and the separation into pure audio data and existing additional data bits takes place. The audio data are transformed into the internal digital format of the d 03 and then reach the digital input selector. The identification bit for the emphasis is evaluated and at the same time the digital deemphasis filter is automatically inserted in the signal path (see also chapter 4.2).
All other additional data (C-Bit, U-Bit) reach the AES output unchanged.
The processing of digital audio data in the consumer format **SP/DIF** or **OPTICAL** is also possible. If signals are present at both the AES/EBU and the SP/DIF inputs at the same time, the AES signal automatically has priority.

The processing of data for the YAMAHA **Y2** and 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 inserted in 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.

The YAMAHA format does not contain any additional control data. The digital deemphasis filter can therefore be switched on manually on the rear panel.

With the digital **input** selection button on the front panel the appropriate input can be selected.

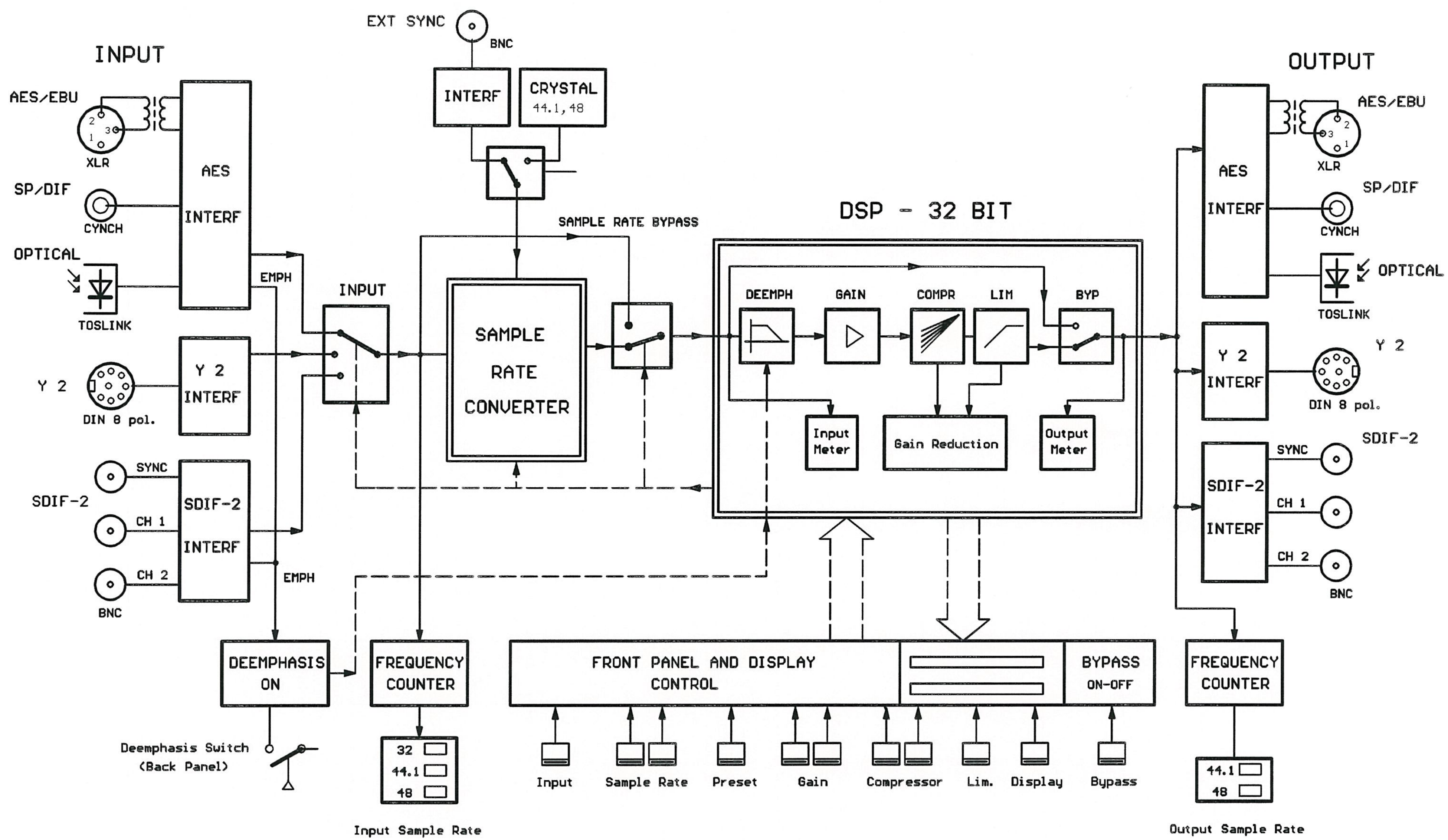


Abb.5 Block Diagram

The **digital sample rate converter** which is integrated in the d 03 processor can be fed into the signal path if necessary.
The conversion of the sample rate is done by an integrated circuit from ANALOG DEVICES The process used in the d 03 for changing the sample rate is called asynchronous sample rate conversion because there does not have to be a relationship between input and output sample rate (see also chapter 1.2).
The resolution of the output signal is limited to 20 bit when the sample rate converter is used.
The source of synchronisation can be internal or external. Internal synchronisation means that the processor d 03 is the master and the sample rate of the output signal is determined by reference clocks inside the device. In this case the sample rate is derived from internal crystal oscillators with a 44.1 or 48 kHz word clock.
With external synchronisation the sample rate is locked to the external word clock applied at the rear panel.
The sample rate converter can also be bypassed and then the output rate is phase-locked with the input rate.

The further processing of the digital audio signal is done in a **Floating-Point Signal Processor** (Texas Instruments) with a data width of **32 bits**. The accuracy of calculation of 32 bits does not lead to a deterioration of the signal quality, even for audio data with 24 bits.
The DSP carries out the functions of the dynamic range processing, the linear gain, the deemphasis filter. It measures the input and output levels and generates the display for the gain reduction. An exchange of data between the function buttons and the display elements of the front panel takes place via a special interface.
(For functions of the individual buttons and LEDs see chapter 2.1).

One main function of the digital dynamics processor d 03 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). The parameter ratio 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 atmo, air-conditioning, hum and noise) may occur during signal pauses.
For that both buttons of ratio had to be pressed at the same time. A red LED is visible In the compressor gain display , which determines the maximum value of compression gain.
This value can be changed with the buttons INC and DEC in the range of 2 dB ... 15 dB.

For the dynamics functions, particularly the algorithm of the **limiter**, an 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.
In the bypass mode the digital signal is fed through unprocessed directly to the output. To avoid phase jitter of the audio signal when switching the bypass on and off, the signal delay time of approx. 2 ms is also active in bypass mode.

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. As all three interfaces are always approached at the same time, the output signal is available in various digital formats in parallel.

4. APPLICATION NOTES

4.1. Presets

Depending on the type of programme signal (genre), it is possible to make an ideal match of the control characteristics of the dynamics processor by selecting the corresponding **preset**. The four presets available are:

1	-	universal
2	-	classicalal
3	-	pop music
4	-	speech

Depending on the preset selected, a complete set of parameters, the attack and release times, the thresholds and the interactions of multiple programme-signal dependent control circuits are changed. Generally speaking e.g. the release times are longest in classical and shortest in speech.
For understanding the basic principle of the Jünger Audio dynamics processors (Multi-loop) see chapter 1.1.
The presets are fixed because with the multitude of parameters and their dependencies specific alterations carried out by the user would be problematic.

4.2. Processing signals containing emphasis

If the audio signal was recorded with **emphasis**, the additional information of the AES/EBU or SDIF-2 input signals contains a definite emphasis-control-bit.
This is sometimes the case in older recordings because the use of emphasis 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 dynamic range 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.

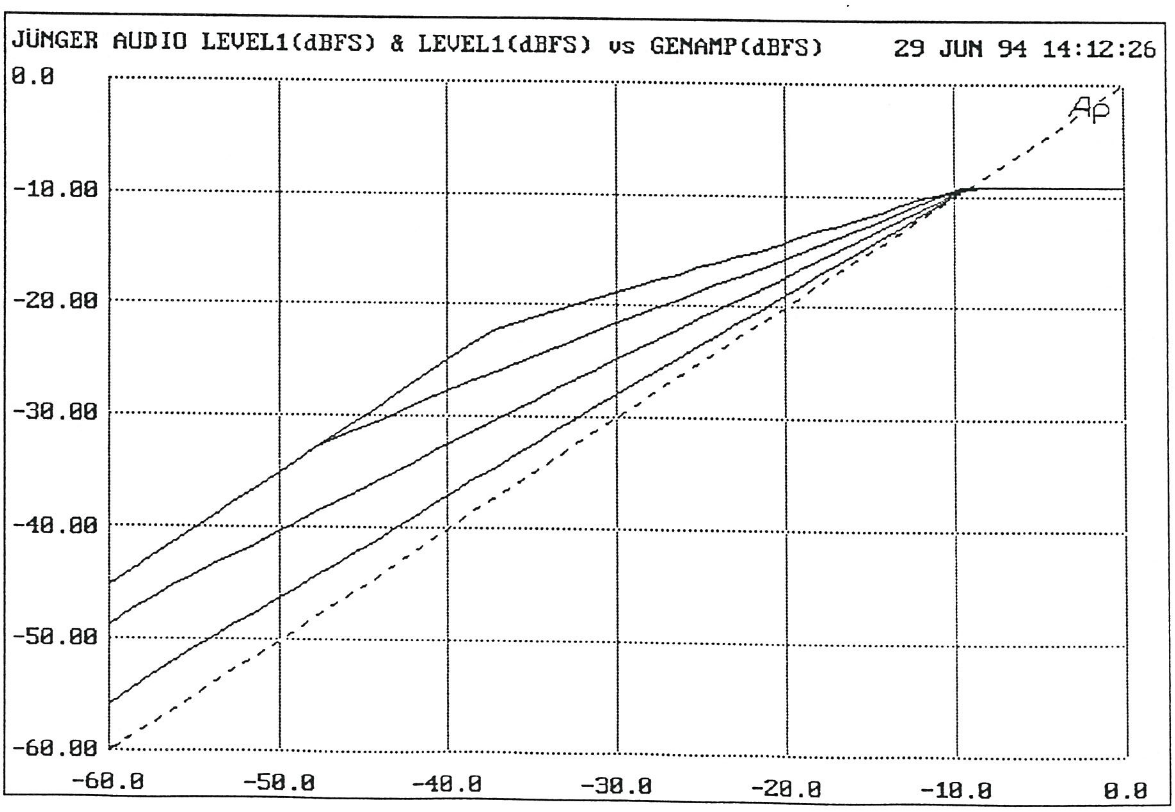
Prior to dynamic processing a signal recorded with emphasis must therefore be linearized, i.e. pass through a digital deemphasis filter.
This filter in the **d 03** is automatically turned on if the corresponding control bit is set in the AES/EBU or SDIF-2 format. The YAMAHA Y2 format does not contain any additional information. In this case the deemphasis filter has to be activated manually with the **DEEMP switch** located on the rear panel.

4.3 Working with headroom

The static characteristics of the processor d 03 (see also figs. 2 and figs. 3) usually refer 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 03 can be calibrated to different digital reference levels. Then the limiter threshold and therefore the maximum output level is the set digital reference level. This value is then also the reference for the compressor and limiter threshold values.
The static characteristics for a reference level of -9 dBFS are illustrated in fig. 6.

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 one can change the digital reference level in the range of 0 dBFS till -15 dBFS (see also chapter 6).

fig. 6



static characteristics, compressor
digital reference level: -9 dBFS
parameter: RATIO

4.4. Influence of the signal delay

The signal delay through the dynamics processor is approx. 2 ms due to the delay of the signal in an internal memory. The signal delay enables a realisation of the algorithms for limiter and compressor with preview time, i.e. to change the signal curve before maximum level is reached.
A problem may occur if a signal that has been processed by the dynamics processor is to be mixed with a direct signal. Because of the different delay times there may be cancellations or superelevations, depending on the phase relation in certain frequencies (comb filter effect). A delay time equalization of both signals must therefore be carried out for a possible mixing.

4.5. Selection of parameters to increase loudness

Signal compression and the loudness enhancement of the digital audio signal evoked by it can be achieved by combining two dynamic range control processes : firstly, the **compression** of 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 shold 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.

4.6. Using the sample rate converter only

The change of the sample rate, following the principle which is described in chapter 1.2. is a calculation process delivering an output signal with a resolution of 20 bit. The accuracy of calculation does not lead to a deterioration of the signal quality for normally used 16 bit digital audio signals. This is valid only for the sample rate converter. The dynamics processor itself has an accuracy of 24 bit.
A problem may occur if the input signal is leveled not far from full scale. Caused by the calculation process output sample values are generated between the original samples which may have a higher level and then may reach full scale (0dBFS). As a result a clip would be detected in the output signal. This phenomenon appears with sample rate converters of several manufacturers.

The only way to avoide clipping is a digital limiter following the sample rate converter.
For applications there the sample rate converter only is used the digital limiter always should be activated .

5. APPLICATIONS

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

- **post production and ADR studios**
  adjusting dynamic range and loudness level of individual takes
  no problems with synchronisation
  free running connections with audio and video equipment

- **FM-Broadcast, TV-Sound**
  signal conditioning
  matching dynamic range of different programme signals
  direct usage of CD-players in 48 kHz systems
  sample rate conversion to 32 kHz

- **digital recording and mixing**
  increased loudness level (compressor, limiter)
  connection of devices without synchronisation
  removing of jitter in digital audio signals (reclocking)

- limiter for **digital transmission links**
  no overloads by application of the digital limiter
  system interconnection with unequal sample rates

further applications without the dynamic functions

- **digital sample rate conversion**
  digital input signal is free running without synchronisation
  output signal with internal or external synchronisation

- **digital audio format conversion**
  all digital outputs are available in parallel
  AES/EBU + SP/DIF + OPTICAL - YAMAHA Y2 - Sony SDIF-2

- **digital deemphasis filter**
  removing emphasis, automatic or manual
  emphasis bit in AES/EBU and SDIF-2 is also removed

6. INSTALLATION

The digital dynamics processor d 03 belongs to the Schutzklasse 1, corresponding to VDE 0804.
It may only be operated using correctly installed power supply systems.

Adjusting the digital reference level

The reference level for the dynamic range processor is the internal digital reference level, which is the maximum output level for the limiter and the reference level for the static compressor characteristics.

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 (gain and preset display are blinking). Pressing the INC or DEC buttons one can change the digital reference level in the range of 0 dBFS till -15 dBFS.

For digital production or transmission the output level should be the maximum, i.e. the digital reference level should be 0dBFS. When working with analog signals 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 digital processing.

7. TECHNICAL SPECIFICATIONS

DIGITAL INPUT / OUTPUT

sampling rate :	30 kHz ... 50 kHz
ratio :	2 : 1...1 : 2 between input and output
audio data format :	24-bit (AES/EBU and Y2)
	20-bit (SDIF-2)
dynamic range :	> 130 dB, (120 dB with sample rate converter)

AES/EU

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

SP/DIF

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

OPTICAL

connector :	TOSLINK
input format :	AES professional, AES consumer
output format :	same as input

Y 2

level :	5 Vpp / 150 Ohm, balanced
connector :	8-pin socket corresponding to DIN 45326
format :	YAMAHA Y2

SDIF - 2

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

power consumption :	approx. 20 W
dimensions :	19 inch, 1 RU, depth 250 mm
weight :	4.5 kg

8. WARRANTY AND SERVICE INFORMATION

JÜNGER AUDIO grants a two-year warranty on the

digital dynamics processor MODEL **d 03**

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

JÜNGER AUDIO - Studiotechnik GmbH

Rudower Chaussee 5 (Geb. 9.51)

D - 12489 Berlin
GERMANY

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

KONFORMITÄTSERKLÄRUNG
DECLARATION OF CONFORMITY

Geräteart: Type of equipment:	Digitaler Dynamikprozessor Digital Dynamics Processor
Produkt / Product:	**model d03**

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: This certification is based on:	Prüfbericht(e) des EMV-Prüflabors 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, (Ort/Place)	02.11.1995 (Datum/Date)	.... (Rechtsgültige Unterschrift/Legally Binding)