DIGITAL TRANSMISSION PROCESSOR

model d05

operation manual

rev. 2.1

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INTRODUCTION

!

The **Digital Transmission Processor model d05** is specially designed for the digital dynamic range processing of audio signals prior to transmission.

The d05 offers dynamic range control which can take into consideration pre-emphasis of the audio signal, performed according to one of the standard pre-emphasis curves (50us, 75us, or CCITT J.17.), which is normally applied to the audio before transmission. With the new Adaptive Spectral Processing developed by Jünger Audio one can achieve more loudness for the transmission than can be achieved using conventional fixed pre-emphasis filters.
The unit has both digital and analog inputs and outputs (analog input via 24bit ADC , analog output via 24bit DAC), performs all dynamic range control and filtering in the digital domain, and offers the same outstanding headroom and audio quality as other models in the family of digital dynamics processors by Jünger Audio.
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 necessary for inaudible processing of the dynamic range are continuously automatically controlled in response to changes in the programme signal.

- fully **digital** processing device
- **stereo link or 2-channel**-mode
- **compressor, limiter**
- **4 programmes** for signal dependent control algorithms
- linear **gain** -6 dB ... +15 dB, in 1 dB steps
- **digital adaptive spectral processing (50 µs, 75 µs)** or **fixed preemphasis filter** (**50µs**, **75µs**, **J.17**)
- multicolour **LED display**
- **AES/EBU digital interface**
- **analog input 24 bit ADC, analog output 24 bit DAC** optional: transformer coupled in- and outputs
- **parameter storable in presets**
- bypass-relais activated by loss of main power
- **key switch** locking the front panel

CONTENTS

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THE DESIGN OF THE DEVICE

1

1.1. basic functions

The digital dynamic range processing of the model d05 digital transmission processor provides **excellent optimisation of programme signal levels** in broadcast applications.

The level of the processed programme signal which is output by the unit can be limited taking into consideration the pre-emphasis which is usually applied in an FM-transmitter system. This prevents over-deviation of the FM signal which could otherwise occur, due to the boosted level of high frequencies in the pre-emphasised programme signal exceeding the maximum permissable signal level for the transmitter.

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 dynamics control of the processor **model d05** can be used in stereo mode or in 2-channel-mode for independent audio signals.

The **digital transmission processor model d 05** changes the dynamic range of digital , as well as analog audio signals. Digital audio signals in the **AES/EBU** standard format can be processed. For the analog inputs high resolution 24 bit A/D-converters are used. The sample rate of the A/D-converter can be synchronised to internal crystal clock generators or to external word clock signals.
The output signals are available in AES/EBU digital format. In addition, an analog signal output is available which operates with 24-bit D/A converters.

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.

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.

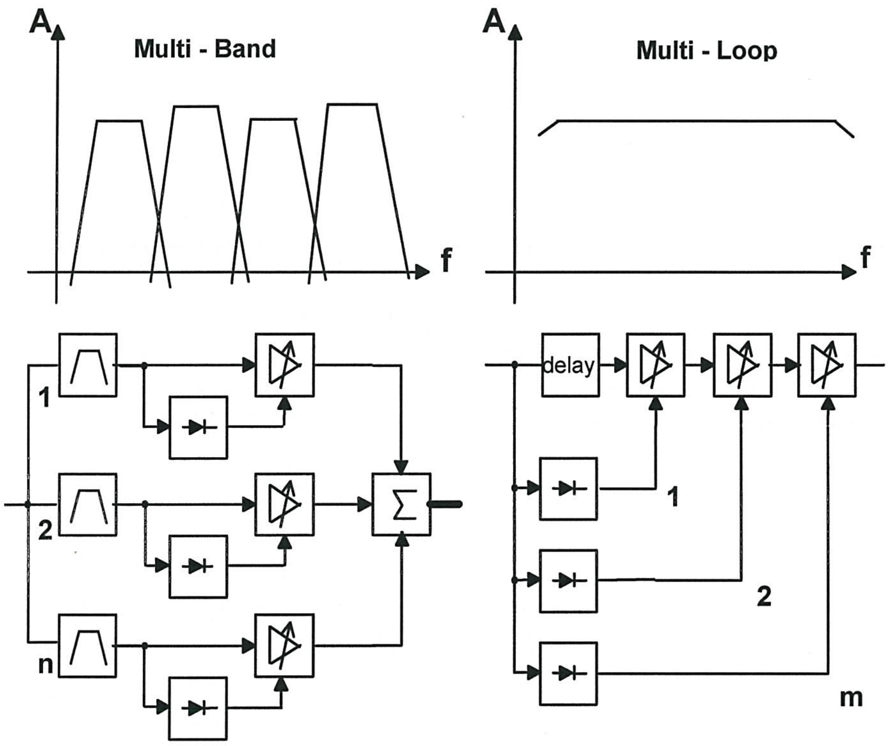


fig. 1:
basic principles of dynamic range processors

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 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 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.

digital dynamics processor

The **d 05 digital transmission processor** is a dynamic range processor which, contrary to its conventional counterparts, is effective for a wide dynamic range of 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.

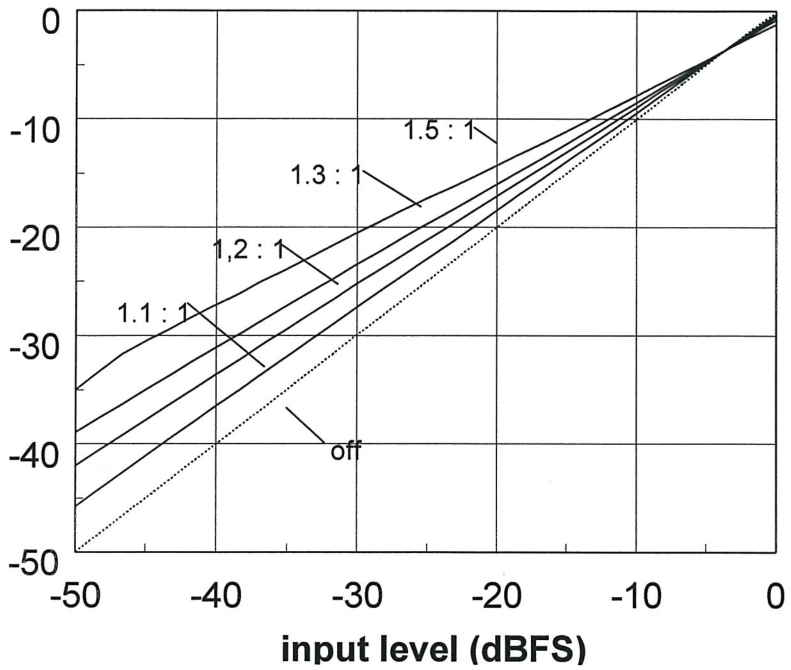
compression

Compression (reduction of the dynamic range of the input signal to match the dynamic range of the FM 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 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.

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).

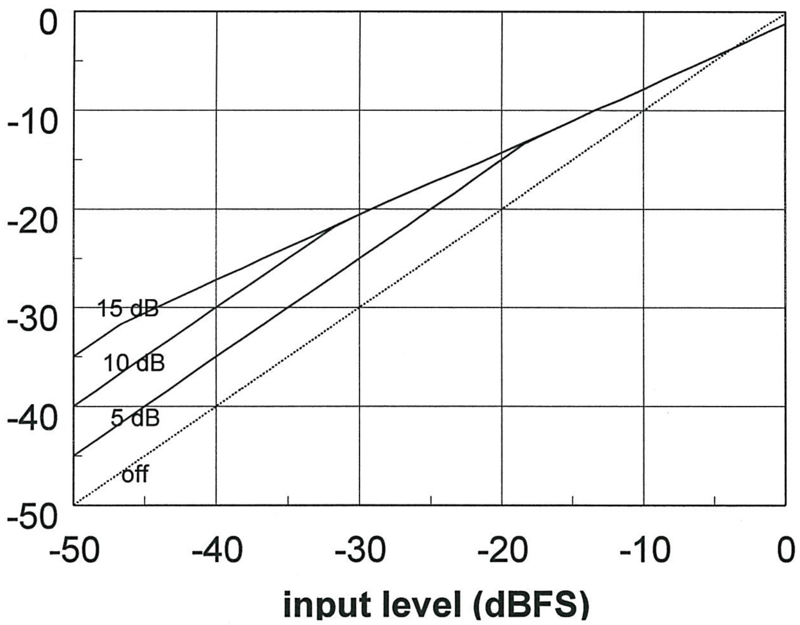
static characteristics: compressor
output level d05 (dBFS), filter off



compression gain: max. 15 dB
parameter: ratio

fig. 2:
static characteristics compressor

static characteristics: compressor
output level d05 (dBFS), filter off



compression gain: max. 15 dB
parameter: compression gain
ratio: 1.5 : 1

fig. 3:
static characteristics compressor

1.3. pre-emphasis filter

To increase the signal to noise ratio for FM transmission pre-emphasis will usually be applied to the audio signal before the transmitter and corresponding de-emphasis applied at the receiver.

The pre-emphasis will increase the level of high frequency components in the transmitted audio signal and therefore increase the amount of deviation in the FM carrier frequency which high frequency components in the audio signal will cause.

If the overall level of the transmission signal before pre-emphasis is not controlled taking into consideration the increase in the level of high frequency signal components, which pre-emphasis will introduce, then there is a danger that high level high frequency content in the signal will cause over-deviation of the FM carrier.

The output signal level of the model d05 digital transmission processor can be limited taking into consideration the pre-emphasis which is usually applied in an FM-transmitter system. The unit offers two different methods for taking pre-emphasis of the audio signal following it's output from the d05 into account.

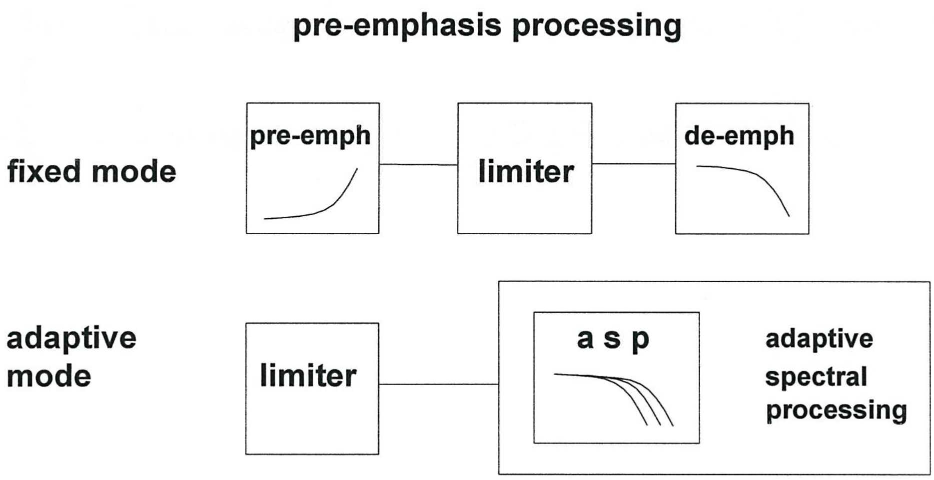


fig. 4: principle function of pre-emphasis control

fixed mode

In **fixed mode** the d05 itself applies pre-emphasis with fixed parameters to the audio signal, before its processing to limit signal level. After the limiter the d05 applies a corresponding de-emphasis to the audio signal. Since the limiter acts on a pre-emphasised signal this ensures that when the audio signal has pre-emphasis applied again at a later point in the signal chain the signal is still limited to a level at which over-deviation in the transmitter will not occur.

The disadvantage of this processing is a lowered maximum level for low and mid frequency parts of the signal since the (broadband) limiter decreases the overall signal level based on the raised level of high frequencies which dominate limiter activity.

This disadvantage can be avoided with the adaptive spectral processing developed by Jünger Audio and used when the unit is in **adaptive mode**.

adaptive mode

In **adaptive mode** the signal does not have pre-emphasis applied before the limiter. Instead the limiter is followed by an additional dynamically controllable low pass filter. The characteristics of this filter are variable and adaptive control of the filters characteristics is used to reduce the level of high frequency signals just at those points in the programme material where the pre-emphasised audio signal would otherwise cause over-deviation in the transmitter.

The reduction in the level of high frequency signals to prevent over-deviation is effective only for very short periods of time at the moments when over-deviation could otherwise occur and very fast release time for this low pass filtering process means that it is, in practice, not audible. The advantage is that signals at lower and mid frequencies are not affected in this case. Output level can be maximised at all times without risk of over-deviation.

INSTALLATION

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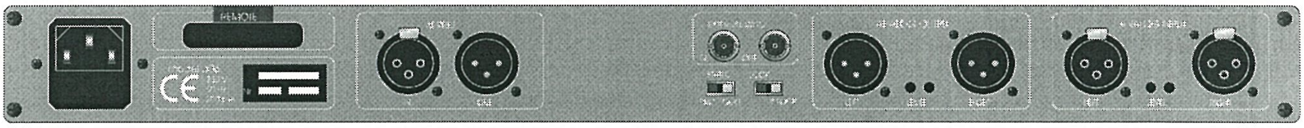
2.1. Power Supply

The digital transmission processor d05 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 transmission processor d05 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 INPUT/OUTPUT

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

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 24 bit 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 sample rate.

Channel status bits are defined in the AES/EBU data stream. With the digital transmission processor d05 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.

LOCK / UNLOCK Switch enables unlocking of key locked units.

LOCK Key on front panel determines locking status of the unit.

UNLOCK Rear panel switch overwrites locking status of front panel key. Unit is unlocked and user has full access for operation.

2.4. Setting the Digital Reference Level

The static characteristics of the processor d 05 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.

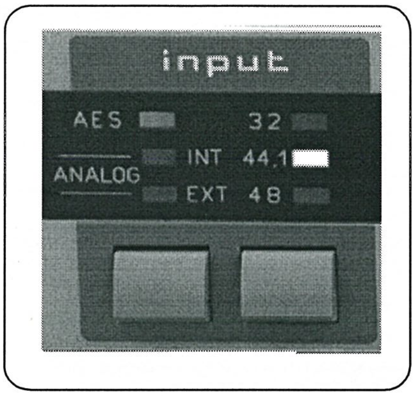
It is possible to store two different digital reference levels, one for use when the analog signal input is selected and another different setting for use when the digital input signal is selected. When changing the input selection between analog and digital the required reference level setting is automatically selected. So it is very easy to optimize levelling and headroom of the model d05 for different applications in analogue or digital mode. **These two adjustments for digital reference level are stored in each of four presets independent!**

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 d05 will output.

OPERATING CONTROLS AND DISPLAYS

3

All functions of the **digital transmission processor d 05** are activated by buttons. The front panel shows easily recognizable function groups.



input

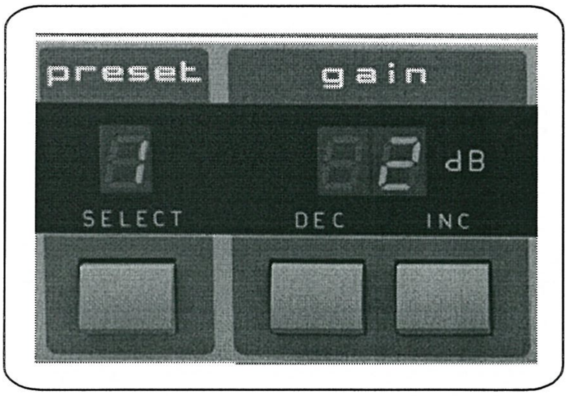
By pressing the left button in the input section the required input signal can be selected. Each time the button is pressed the input selection is changed and one of the three LED's above the button lights to show the newly selected input.

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

All parameters adjustable on the front panel of the unit can be stored as a parameter set in a preset.

Four different sets of all parameter settings (four different preset settings) can be stored.

Recalling a previously stored preset containing all parameter settings for a particular type of programme material allows all settings of d05 to be configured for a particular type of programme material very quickly.

Recalling a previously stored preset can be done either locally, by pressing the SELECT button on the front panel repeatedly until the desired preset number is displayed, or remotely, via the connection to the rear panel REMOTE connector (parallel or RS232) if the remote option is installed.

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.

pre-emphasis

The model d05 digital transmission processor offers dynamic range control which takes into account the application of pre-emphasis (50µS, 75µS or CCITT J.17 type pre-emphasis) to the audio signal prior to transmission, after it has been output from the d05. It is possible to select between fixed mode and adaptive mode (see section 1.2) for handling pre-emphasis by pressing the left button in the pre-emphasis section.

With the right button one can select the type of pre-emphasis (50µS, 75µS or CCITT J.17 type) which will be applied during transmission and which should therefore be taken into account during the dynamic range control processing. The LED's show the selected pre-emphasis type and filter mode.

The ADAPT-LED lits red everytime when the adaptive filter is active (only in adaptive filter mode)**.**

Note CCITT J.17 type pre-emphasis can only be handled in the fixed mode.

compressor/ limiter

Press the PROGRAM 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 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.

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

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, 1.2:1, 1.3:1, or 1.5:1). 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 gain

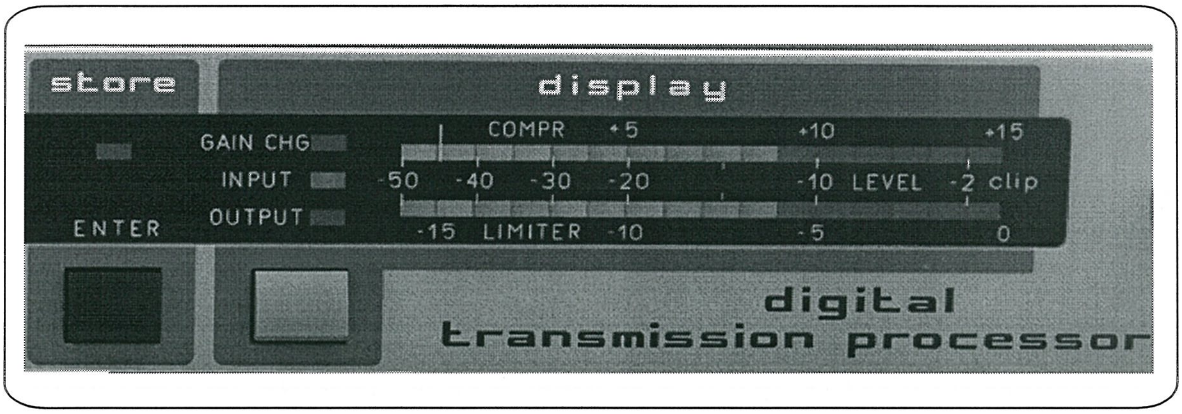
Compression is partly achieved by increasing the level of low level signals, (the lowest of which might otherwise be below the noise floor of the FM transmission system). The lower the input signal level the higher the additional gain applied to that input signal by the compression processing will be. The **maximum amount of gain** applied to a low level signal can be adjusted independently of the compression ratio. Press both the PROGRAM button and the RATIO button at the same time.
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.

digital reference level

The limiter limits the maximum output signal level of the d05 precisely to the set **digital reference level**. (see also 5.3, and, for details of setting the digital reference level, see under "display" on page 18). The limiter is always active to ensure that output level of the d05 never exceeds the preset digital reference level.Two different reference levels can be set, one reference level for use when using a digital input signal, and another for use when using an analog input signal.

link

Press the LINK button to switch between two channel mode and stereo mode.When the red LINK LED is lit the unit is in stereo mode. Processing of the two input channels is linked - both channels are controlled in the same way for processing a stereo signal ensuring that the original stereo imaging is maintained (no shift in stereo balance occurs due to processing). When the LINK LED is not lit the two channels work independently of each other allowing simultaneous processing of two independent mono signals.



store

A first press of the ENTER button in the store section prepares the unit to store a preset. The unit goes into a "ready to store" mode and the store LED and the preset number display will flash on and off continuously. At this stage all parameters, including the preset number to store to, can be changed by the user. Then pressing the ENTER button for a second time will actually store the units current settings into the currently selected preset.

If you do not press the ENTER key for a second time then, after a short time the "ready to store" mode is cancelled and store LED and preset display stop flashing.

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).

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.

The key locking switch enables prevention of unauthorized adjustments during On-Air operation of the unit. With the key removed no change of parameters is possible - only use of the **DISPLAY** key for changing display mode is possible. (see also Chapter 2.3 LOCK/UNLOCK switch!)

lock

FUNCTIONAL DESCRIPTION

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After switching the power on, the **digital transmission processor d05** automatically chooses the settings used before the power was turned off.
All parameters used, e.g. input, preset, gain, program, compressor, link and display, are stored and re-applied.

digital input signal

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 d05 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.
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 d05 can synchronise to properly.

AES/EBU input signal

If AES digital input is selected but the d05 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 d05 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 d05'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 d05'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.

analog input signals

Analog audio input signals can be fed into the unit via the analog input XLR connectors and first pass through an electronically balanced (optional transformer coupled) 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 0dBu 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.

24 Bit ADC

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 AKM Semiconductor. The analogue to digital converter has a dynamic range of 110dB 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 d05's internal digital format for processing.
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 d05'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.

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 d05) will be indicated by the lighting of a yellow LED in the d05's front panel sampling frequency display.

synchronization

All three LED's in the d05's front panel sampling frequency display will flash red if the d05 can not synchronise to an extrnal 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).

DSP functions

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 2.1).

compression

One main task of the digital transmission processor d 05 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 1.5:1. The transition to another characteristic can be carried out during the running programme. It does not cause any clicking noises.

maximum compression gain

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.

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.

look ahead limiter

signal delay

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.

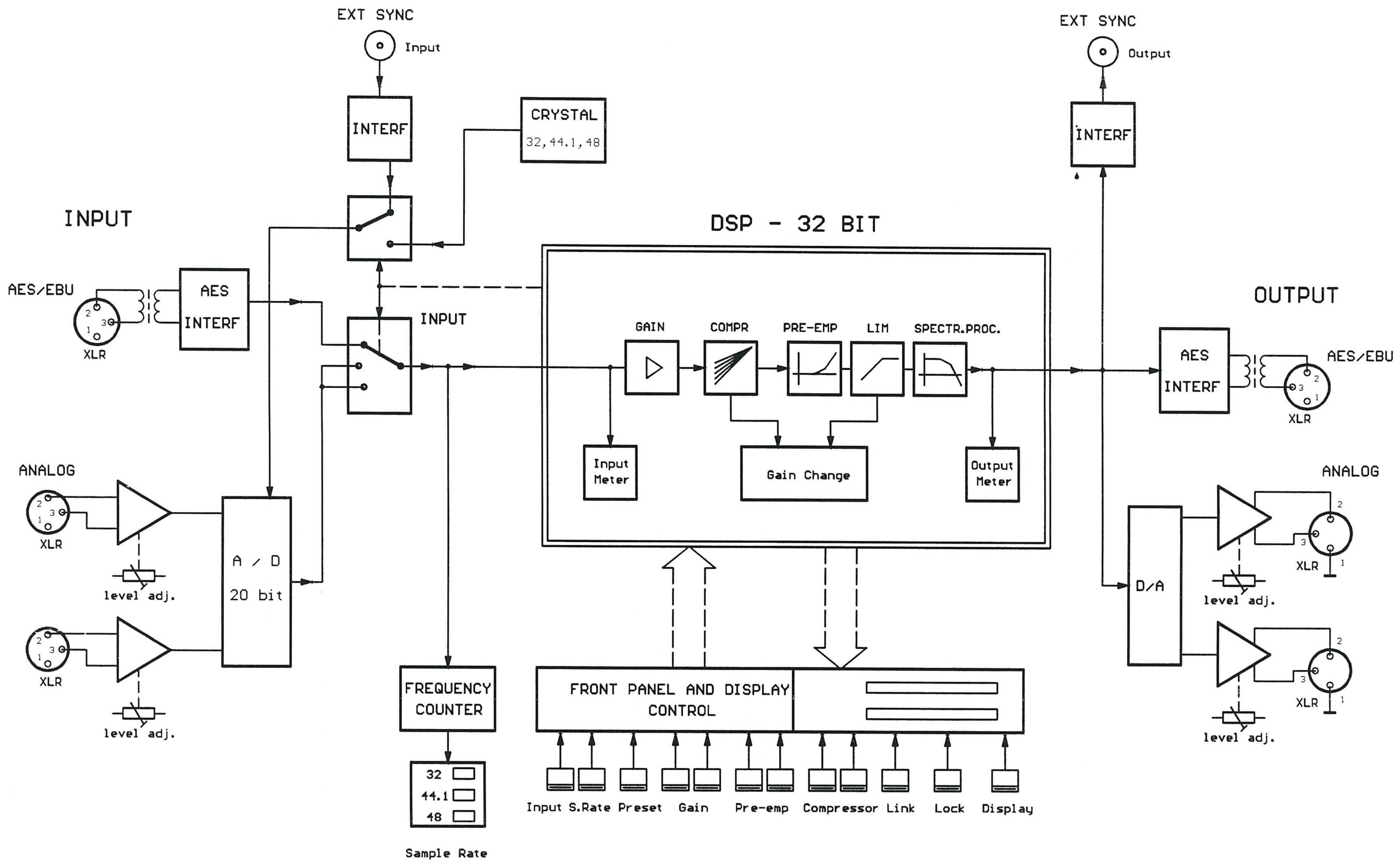
analog output

Additionally, an analog output signal is available. A stereo - D/A converter with a resolution of 24 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 0dBu 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 (optional transformer coupled) analog output drivers is such that the output level is maintained even when driving an unbalanced load.

The block diagram of the signal processing in the **d 05** is shown in fig. 5.



Block Diagram d 05

APPLICATION NOTES

5

5.1. Programmes

It is possible to choose one of four 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:

	Program	range of adaptive processing time
1 -	universal	70 ms to 5.0 s
2 -	pop music	30 ms to 2.5 s
3 -	speech	15 ms to 1.2 s
4 -	live	2 ms to 0.2 s

The basic Multi-loop principle of the Jünger Audio dynamics processors operates with adaption of dynamic range control parameters to the incoming audio signal . That means permanently analysis and calculation of attack times, release times, thresholds and interaction parameters of several frequency linear control circuits. Inside the range of processing time, the time constants can be varied by the adaptive processing.
(please refer to chapter 1.1 also).

Selecting a particular preset sets up the optimum parameters of the dynamics processor for a particular kind of programme material. For example the new LIVE program will be helpful for all material with public audience (applause, voices and more).

Fixed presets are used because of the great number of parameters and the interaction of parameters in different stages of the multiloop system, changing of parameters by the user could cause problems.

(Note: However it will be possible to load any future improvements to the fixed presets in the form of software updates easily into existing d05 units. Downloading is made by loading new software into the d05 over the AES/EBU digital interface).

5.2. 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.

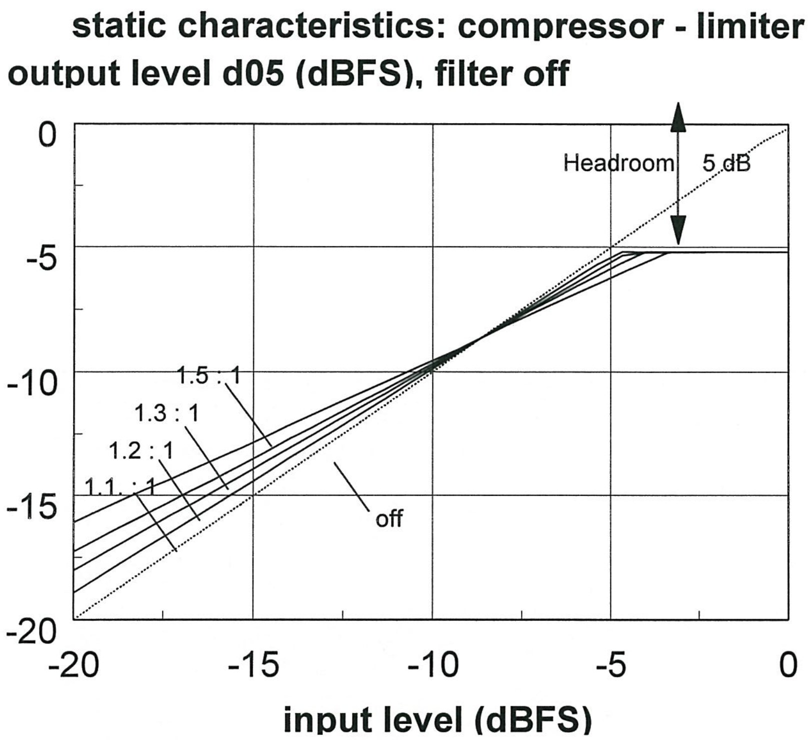
5.3. Adjustment of the digital reference level

The static characteristics of the processor d 05 are illustrated in fig. 6.

These characteristics are measured using the digital input and output (measured in dBFS). Working with the analogue inputs and outputs one has to consider the calibration of the ADC and the DAC.

With the factory setup the unit is calibrated to the nominal level of +6 dBu, e.g. an analogue signal level of +6 dBu is corresponding to the internal digital level of -9 dBFS.

The reference level for the dynamic range processor is the internal 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 4 dB under the internal digital reference level.



compression gain: max. 15 dB
parameter: ratio
digital reference level: -5 dBFS

fig. 6: static characteristics:

compressor / limiter with -5 dBFS 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 d05 for different applications in analogue or digital mode.

For a digital 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 d05 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 transmitter 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 d05 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 d05 will output.

5.4. Influence of the 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 pages 4 and 5).
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.

APPLICATIONS

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- **FM-Broadcast, TV-Sound**
  transmission signal conditioning
  matching dynamic range of different programme signals
  increasing signal loudness level
  maximized modulation without risk of over-deviation

- limiter for **digital or analog transmission links**
  no overload by application of the digital limiter

- **more dynamic range** for CCITT J.17 transmission links
  digital full scale without clipping
  optimized level without headroom for J.17 transmission links

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

applications

further applications without the dynamic functions

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

TECHNICAL SPECIFICATIONS

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

digital input / output

AES/EBU

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

analog input / output

A/D converter :	stereo, 24 Bit, oversampling
dynamic range :	110 dB (RMS)
input level :	+6...+22 dBu for 0 dBFS, adjustable
input :	XLR, floating balanced, 20 kOhm (optional: transformer balanced)
D/A converter :	stereo, 24-bit, oversampling
dynamic range :	108 dB (RMS)
output level :	+12...+22 dBu for 0 dBFS, adjustable
output :	XLR, floating balanced, 50 Ohm (optional: transformer balanced)

filters

fixed :	standard pre-emphasis 50 µs, 75 µs, CCITT J.17
variable :	adaptive spectral processing 50 µs, 75 µs

general

remote :	for connection with remote via drc05 remote interface board (optional)
power consumption :	approx. 20 W
dimensions :	19 inch, 1 U, 250 mm depth
weight :	5 kg

WARRANTY AND SERVICE INFORMATION

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JÜNGER AUDIO grants a two-year warranty on the

digital transmission processor MODEL **d05**

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

JÜNGER AUDIO - Studiotechnik GmbH

Justus-von-Liebig-Strasse 7

D - 12489 Berlin
GERMANY

Tel.: (*49) -30-677721 - 0
Fax.: (*49) -30-677721 - 46
e-mail: sales@junger-audio.com

KONFORMITÄTSERKLÄRUNG
DECLARATION OF CONFORMITY

Geräteart: Type of equipment:	Digitaler Übertragungsprozessor Digital Transmission Processor
Produkt / Product:	**model d05**

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)